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Towards adaptation of OFDM based wireless communication systems

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Towards Adaptation of OFDM Based Wireless Communication Systems

by

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DEDICATION

To my family

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LIST OF ACRONYMS

ADSL Asymmetric Digital Subscriber Line

AR Auto-Regressive

AQAM Adaptive Quadrature Amplitude Modulation

A/D Analog to Digital

AWGN Additive White Gaussian Noise

BER Bit Error Rate

BPS Bits Per Symbol

BPSK Binary Phase Shift Keying

CDMA Code Division Multiple Access

CINR Colored Noise to White Noise Ratio

CIR Channel Impulse Response

DAB Digital Audio Broadcasting

DAC Digital to Analog Converter

D/A Digital to Analog

DER Detection Error Rate

DFE Decision Feedback Equalizer

DFSE Decision Feedback Sequence Estimation

DFT Discrete Fourier Transform

DSL Digital Subscriber Line

DSPs Digital Signal Processors

DVB-T Terrestrial Digital Video Broadcasting

FDM Frequency Division Multiplexing

FFT Fast Fourier Transform

FPGA Field-Programmable Gate Array

HDSL High Speed Digital Subscriber Line

HyperLAN High Performance Local Area Network

ICI Inter-carrier Interference

IDFT Inverse Discrete Fourier Transform

IEEE Institute of Electrical and Electronics Engineers

IFFT Inverse Fast Fourier Transform

IIR Infinite Impulse Response

ISI Inter-symbol Interference

K-L Kullback Leibler

LAN Local Area Network

LMMSE Linear Minimum Mean-square Error

MC-CDMA Multi-carrier CDMA

ML Maximum Likelihood

MMSE Minimum Mean-square Error

MLSE Maximum Likelihood Sequence Equalization

MSE Mean-squared-error

OFDM Orthogonal Frequency Division Multiplexing

PA Power Amplifier

PAPR Peak-to-Average Power Ratio

PDF Probability Distribution Function

PSD Power Spectral Density

PSK Phase Shift Keying

QAM Quadrature Amplitude Modulation

QPSK Quadrature Phase Shift Keying

QoS Quality of Service

RF Radio Frequency

RMS Root-mean-squared

RSSE Recursive Soft Sequential Estimation

SNR Signal-to-noise Ratio

TDD Time Division Duplexed

TDMA Time Division Multiple Access

VLSI Very Large Scale Integrated Circuits

WLAN Wireless Local Area Network

WMAN Wireless Metropolitan Area Network

WSSUS Wide-sense Stationary Uncorrelated Scattering

TOWARDS ADAPTATION OF OFDM BASED WIRELESS COMMUNICATION SYSTEMS

Sharath Reddy Billoori

ABSTRACT

OFDM has been recognized as a powerful multi-carrier modulation technique that provides efficient spectral utilization and resilience to frequency selective fading channels. Adaptive modulation is a concept whereby the modulation modes are dynamically changed based on the perceived instantaneous channel conditions. In conjunction with OFDM systems, adaptive modulation is a very powerful technique to combat the frequency selective nature of mobile channels, while simultaneously attempting to fully maximize the time-varying capacity of the channel. This is based on the fact that frequency selective fading affects the sub-carriers unevenly, causing some of them to fade more severely than others. The modulation modes are adaptively selected on the sub-carriers depending on the amount of fading, to maximize throughput and improve the overall BER.

Transmission parameter adaptation is the response of the transmitter to the time-varying channel quality. To efficiently react to the dynamic nature of the channel, adaptive OFDM systems rely on efficient algorithms in three key areas namely, channel quality estimation, transmission parameter selection and signaling or blind detection mechanisms of the modified parameters. These are together termed as the enabling techniques that contribute to the effective performance of adaptive OFDM systems.

This thesis deals with higher performance and efficient enabling parameter estimation algorithms that further improve the overall performance of adaptive OFDM systems. Traditional estimation of channel quality indicators, such as noise power and SNR, assume that the noise has a flat power spectral density within the transmission band of the OFDM signal. Hence, a single estimate of the noise power is obtained by averaging the instanta-

neous noise power values across all the sub-carriers. In reality, the noise within the OFDM bandwidth is a combination of white and correlated noise components, and has an uneven affect across the sub-carriers. It is this fact that has motivated the proposal of a windowing approach for noise power estimation. Windowing provides many local estimates of the dynamic noise statistics and allows better noise tracking across the OFDM transmission band. This method is particularly useful for better resource utilization and improved performance in sub-band adaptive modulation, where adaptation is performed on the sub-carriers on a group-by-group basis based on the observed channel conditions.

Blind modulation mode detection is another relatively unexplored issue in regard to adaptation of OFDM systems. The receiver has to be informed of the appropriate modulation modes used at the transmitter for proper demodulation. If this can be done without any explicit signaling information embedded within the OFDM symbol, it has the advantage of improved throughput and data capacity. A model selection approach is taken, and a novel statistical blind modulation detection method based on the Kullback-Leibler (K-L) distance is proposed. This algorithm takes into account the distribution of the Euclidian distances from the received noisy samples on the complex plane to the closest legitimate constellation points of all the modulation modes used.

CHAPTER 1

INTRODUCTION

Modern wireless communication systems have come a long way since Marconi first demonstrated their ability in 1897. The contemporary definition of wireless communication is the ability to access information ubiquitously without the need of a fixed cable connection. In modern times, wireless applications are moving from the traditional voice-centric to multimedia-centric applications, thereby pushing the need for higher data-rates to support these bandwidth hungry applications. Consequently, enabling wireless technologies also need to continually evolve to sustain this growth.

A major limiting factor in designing high data rate systems is the detrimental effect of the wireless channel. During the propagation of the signal through the channel, it suffers rapid fluctuation in amplitude and phase, and experiences multi-path signals due to different delays caused by reflections of the signal. For most practical mobile channels, where the signal propagation takes place in an atmosphere close to the ground and through many obstacles, a signal can travel from the transmitter to the receiver over multiple reflective paths. These reflected components add up either constructively or destructively at the receiver, and the resulting effect is termed frequency selective fading. In other words, frequency selective fading occurs when the channel introduces time dispersion causing its delay spread to exceed the symbol period. As the data rate increases, the symbol period decreases thereby making the effect of frequency selective fading more pronounced. It is very important to combat the effects of frequency selective fading because the amount of fading is directly related to the throughput, and ultimately the Quality of Service (QoS) of the system. Multi-carrier modulation is a method to combat the detrimental effects of frequency selective channels by transmitting data over multiple parallel channels. One such multi-carrier modulation technique that has gained popularity over recent years is Orthogonal Frequency Division Multiplexing (OFDM). The approach used here is to divide a single high data-rate stream into several parallel lower data-rate streams of data, thereby reducing the data-rate in the individual carriers significantly, while still maintaining overall throughput. The symbol duration in each of the sub-carriers increases, thus eliminating the need for equalization. Hence the goal of increasing the symbol duration is achieved without sacrificing the data-rate, while simultaneously providing immunity against frequency selective fading. In the past, OFDM applications have been scarce because of the complexity involved in practical implementation. Recent advances in technology encompassing Very Large Scale Integrated Circuits (VLSI) and Digital Signal Processors (DSPs) have enabled the cost-effective and practical implementation of the (Discrete Fourier Transform) DFT / Inverse Discrete Fourier Transform (IDFT) via the Fast Fourier Transform (FFT) / Inverse Fast Fourier Transform (IFFT) operation on a single chip and have spurred an array of applications such as their adoption as the next European digital audio broadcasting standard (DAB) [1] and for Terrestrial Video Broadcasting (DVB-T) systems. OFDM is also employed for fixed wireline applications such as Asynchronous Digital Subscriber Line (ADSL) [2], and High bit-rate Digital Subscriber Line Systems (HDSL) [3]. The standardized High PERformance Local Area Network standard known as HIPERLAN/2 in Europe and the IEEE 802.11a/g, that were both designed for indoor wireless networking also incorporate OFDM. More recently, the 802.16a standard, amended by the IEEE as a WMAN networking protocol, uses OFDM in the physical layer specifications. It is aimed at enabling a wireless alternative for cable, DSL and T1 services for last-mile broadband access. OFDM is also a forerunner in the technology choice for 4G mobile systems.

Frequency and time domain fading are major performance limiting factors in wireless systems. Over a dispersive channel, severe signal-to-noise ratio (SNR) fluctuations occur, which points to the fact that fixed-mode transceivers cannot perform effectively without a trade-off in design complexity or throughput. It is this drawback that been a motivating factor for research in the past decade on variable-mode, or adaptive transceivers. The idea is to maximize the available capacity of the channel by adaptively changing the transmission parameters according to the observed instantaneous channel conditions. Adaptation in conjunction with OFDM, termed adaptive OFDM, is a powerful method to mitigate the frequency selective fading nature and utilize the maximum available capacity of the time varying channel. Unlike adaptive single-carrier systems, adaptive OFDM systems have the capacity to react to the frequency selective nature of the channel, in both time and frequency domain. A number of existing standards already incorporate some degree of adaptivity, and further research is likely to be in the direction of performance improvement and merging them into existing standards.

The performance of adaptive OFDM systems depends on the effectiveness of techniques used in the areas of channel quality estimation, transmission parameter adaptation, and signaling or blind detection of these modified parameters. Effective channel quality estimation is vital in selecting the optimal transmission parameters for the next OFDM burst, and efficient signaling or blind detection schemes would reduce or even eliminate the need to transmit information of the modified parameters to the receiver. This thesis will focus on high performance and practical algorithms that facilitate efficient adaptation of OFDM based wireless communication systems.

In the past, estimation of the channel quality indicators, particularly noise power and SNR estimation, has been approached with the assumption that the noise is white and Gaussian distributed. Hence, a single estimate of the noise power is obtained by averaging the instantaneous noise power values across all the OFDM sub-carriers. In reality, the interfering noise is made up of white as well as correlated or colored noise. Color of the noise is reflected by its uneven spectral variation. Typical sources of colored noise include cordless phones, baby monitors, Bluetooth devices other OFDM symbols and any other interfering source that is sharing part of the transmission bandwidth with the OFDM symbol. Using conventional methods to average the noise power across the OFDM spectrum would result in bit errors occurring on those sub-carriers that are severely degraded. Therefore, a windowing approach that takes into account the spectral variation of the colored noise as well as the variance of the white noise is adopted to estimate the noise power[1]. When the noise is completely white, the window size needs to be maximum to obtain the least MSE between the true and estimated noise variance. Similarly, when the noise is completely colored, the smallest window size provides optimal tracking the dynamic noise power across the

[1]This work is partly published in [4]

OFDM band. In scenarios where both components are present, a trade-off between the both extremes needs to be obtained to provide an optimal window size. The optimal window size for a mixture of colored and white noise under different colored-interference-to-white-noise ratio (CINR) and based on the frequency correlation of colored interference is derived[2]. The colored noise is modeled as an auto-regressive (AR) process in time domain, whose order is an indication of the degree of correlation or color of the interfering noise. The proposed method provides many local estimates of the instantaneous noise power across the OFDM spectrum, thereby allowing better tracking of the non-constant noise statistics. This is particularly useful in sub-band adaptive modulation where the sub-carriers are grouped together and adaptation is done on the entire sub-carrier group. In situations where the noise is completely white, the proposed scheme works as well as the conventional algorithms, thereby providing a robust approach to noise power estimation.

Another area in adaptation that has been investigated is blind modulation mode detection. After adaptation of the modulation mode is performed based on the perceived channel conditions, the receiver has to be informed as to which modulation mode to be used for the detection of the transmitted OFDM symbols. This information can be conveyed to the receiver either by means of explicit signaling data transmitted on reserved sub-carriers, or detected blindly by means of blind modulation mode detection mechanisms. Blind modulation mode detection is more beneficial because it eliminates the need for additional signaling data, resulting in increased throughput and data capacity. In order to detect the modulation scheme, the empirical data provided by the noisy data *i.e.* the Euclidian distance to the closest legitimate modulation mode symbol must be used efficiently. Minimization of this Euclidian distance metric to detect the modulation mode has been proposed in [5]. This algorithm fails to perform well in cases where the channel SNR is low. This is because minimization of Euclidian distance at low SNR values would always point to the highest modulation order due to the fact that it has the most number of legitimate constellation points. An improved statistical blind modulation detection scheme which removes this bias toward the higher modulation schemes is proposed. The proposed technique incorporates the distribution of these Euclidian distances into the decision making process. Instead of

[2] This work has been submitted to the IEEE Radio & Wireless Conference (RAWCON 2004)

minimizing only the Euclidian distance, a model selection approach is taken to minimize the Kullback-Leibler (K-L) distance between the obtained Euclidian distance probability distribution and a set of known candidate probability distributions[3]. Adopting this method eliminates the bias towards the higher order modulation schemes and performs well at low channel SNR values, while still maintaining a low level of computational complexity.

1.1 Organization of Thesis

This thesis comprises of six chapters. Chapter 2 describes the building blocks, operation and signal processing aspects of a typical OFDM based wireless communication system. The advantages, disadvantages and applications are also presented. Chapter 3 outlines the principles and practices behind adaptation in the context of OFDM systems. A comprehensive literature review is given along with mention of the enabling techniques that facilitate efficient adaptation. Noise power and SNR estimation is dealt with in Chapter 4, and a novel algorithm that is specifically suited to sub-band adaptive OFDM systems is presented. This algorithm removes the assumption that the interfering noise is white and Gaussian distributed, and takes into account the variation in noise power across the OFDM sub-carriers. Chapter 5 addresses the issue of blind modulation mode detection. A model selection approach that uses the Kullback-Leibler (K-L) distance to detect the modulation mode for use in adaptive and sub-band adaptive OFDM systems is proposed. Conclusions and topics for further research are given in Chapter 6.

[3] This work is partly published in [6] and it is currently under review for another publication [7].

CHAPTER 2

AN OVERVIEW OF OFDM

In multi-carrier modulation systems, the transmission bandwidth is divided into several narrow sub-channels and data is transmitted in parallel on these sub-channels on different frequencies or sub-carriers. Data in each sub-carrier is modulated at a relatively low rate so that the delay spread of the channel does not cause any degradation of the transmitted signal. The result is that each of the sub-carriers will experience a flat response in frequency. OFDM is a multi-carrier modulation technique that divides a communications channel into a number of equally spaced and overlapping sub-carriers. Each sub-carrier is orthogonal (independent of each other) with every other sub-carrier, differentiating OFDM from the commonly used Frequency Division Multiplexing (FDM). This orthogonality between sub-carriers prevents the demodulators from seeing frequencies other than their own. The benefits of OFDM are high spectral efficiency, and most importantly lower multi-path distortion. This is useful in typical terrestrial scenarios because there are multiple time delayed versions of the transmitted signal that arrive at the receiver via paths of different length. Since multiple versions of the signal interfere with each other it becomes very hard to extract the original information. OFDM can overcome this time dispersion induced by the channel, which is a major limiting factor in high data rate communications.

This chapter begins by briefly explaining the channel induced problems in high-speed wireless communications. The concept of OFDM and a system level description of a typical OFDM transceiver is presented, followed by its merits, shortcomings, and applications. For additional details concerning the operation of OFDM transceivers apart from what is presented, the reader is urged to refer [8, 9].

2.1 Multi-path Channels

During the propagation of the signal through the wireless channel, it suffers rapid fluctuation in both amplitude and phase. Flat fading occurs when the channel has a constant gain over a bandwidth that is greater than the bandwidth of the transmitted signal *i.e.*, $B_s < B_c$ or $\tau_m < T_s$, where T_s and B_s the symbol period and transmission bandwidth respectively and τ_m and B_c are the maximum excess delay and coherence bandwidth of the channel respectively. As the need for high data-rates increases, the transmitted symbol durations become smaller and hence *frequency selective fading* poses as a major system design problem. The time dispersion induced because of the signal traveling via multiple paths causes it to undergo frequency selective fading. It is caused when the gain of the channel is not constant over the transmission bandwidth of the signal *i.e.*, $B_s > B_c$ or $\tau_m > T_s$. This is illustrated in Fig. 2.1.. As a result, the received signal comprises of many versions of the transmitted signal which are delayed in time. The resulting effect is termed as Inter-Symbol Interference (ISI). In the frequency domain, certain sub-carrier frequencies are observed to have greater gains than others.

Every propagation path can be characterized by a fixed delay τ_r and a time varying amplitude $A_r(t)$. The impulse response of the channel can be modeled mathematically by the ensemble of all the propagation paths as,

$$\boxed{h(\tau, t) = \sum_{r=1}^{R} A_r(t)\delta\big(t - \tau_r(t)\big)} \tag{2.1}$$

where r represents the echoes arriving at the receiver. Maximum excess delay is the delay between the first and last significant echo *i.e.*, $\tau_r(t) - \tau_1(t)$.

2.2 Introduction to OFDM

2.2.1 Motivation and Applications

Given the pace at which high-speed wireless communications is growing, the technology required to sustain this growth has to continually evolve to sustain this growth. High data-rates require the symbol time of the transmitted data be short. When the symbol time

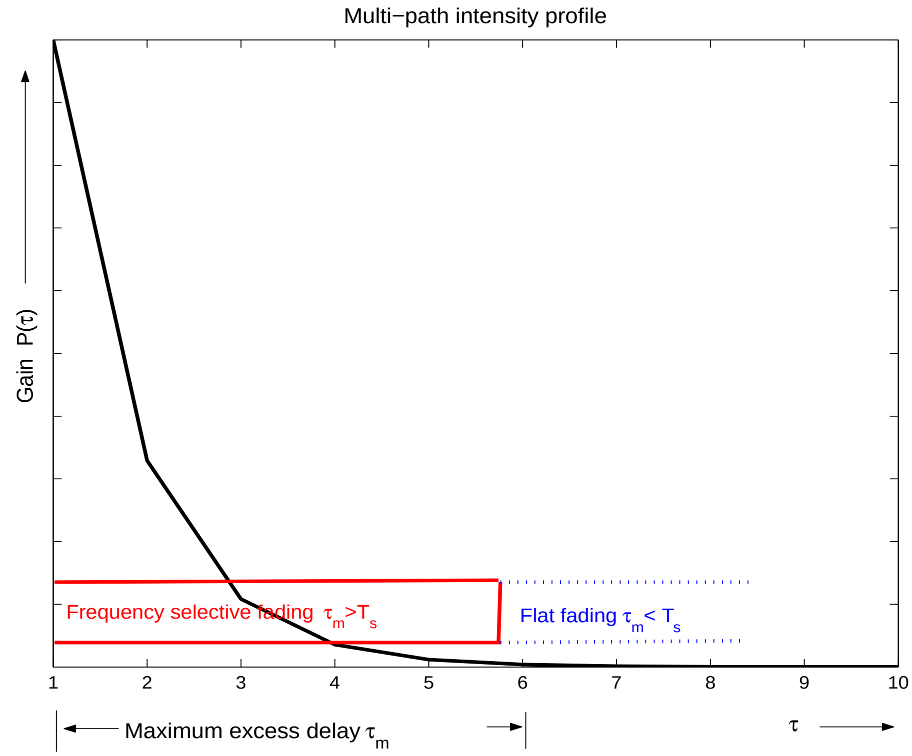

Figure 2.1. A Typical Channel Impulse Response.

becomes shorter than the maximum excess delay of the channel, the effect of ISI becomes a major problem that has to be suppressed by some advanced receiver structures. The signal received via the second path acts purely as interference, since it carries information belonging to a previous symbol or symbols, as illustrated in Fig. 2.2.. Equalization is the standard method to combat ISI in a single-carrier systems. The common equalizers used in contemporary wireless systems are Maximum Likelihood Sequence Equalizers (MLSE), Decision Feedback Equalizers (DFE), Decision Feedback Sequence Estimation (DFSE) and Recursive Soft Sequential Estimation (RSSE). Increased data rates would shorten the symbol time even more leading to ISI spanning several symbols. Complex equalizer structures are needed to handle such severe ISI scenarios, which is not practical in some cases. To cope with any appreciable level of delayed signals, the symbol rate must be reduced sufficiently so that the total delay spread is only a fraction of the symbol period. Thus, the information carried by a single-carrier system is limited in the presence of multi-path.

OFDM has the inherent capability to get around the this problem. It is achieved by dividing the high data-rate stream in several parallel low data-rate streams, and transmitting

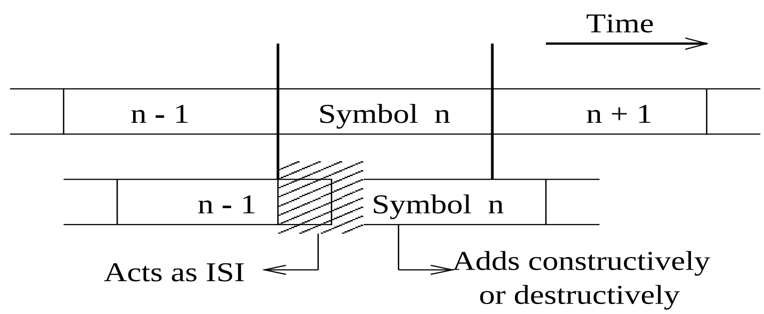

Figure 2.2. Illustration of ISI due to Multi-path.

each of them on separate carrier frequencies or sub-carriers. This has the effect of increasing the symbol duration on each sub-carrier, thereby suppressing ISI originating from several symbols and simplifying the equalization process, while simultaneously maintaining high data-rates. The equalization process at the receiver can be even eliminated if the symbol duration is made longer the maximum excess delay of the channel. This can be made possible by artificially prolonging the symbol and by introducing a guard band, which will be explained in subsequent sections.

In summary, it is very important to combat the detrimental effects of frequency selective fading in mobile environments because the amount of fading is directly related to the throughput and performance of wireless systems. Different methods that have been adopted for differnt systems to mitigate frequency selective fading such as employing DFE [10], MLSE [11], spread spectrum techniques, etc. OFDM is gaining tremendous momentum in the recent past because of improvements in VLSI technology and DSPs which allows efficient and cost-effective implementation of the FFT and IFFT operation. The basic idea of OFDM dates back to the 1950's and was originally referred to as *Kineplex*. It has now been implemented in numerous standards including the recently popular IEEE 802.11a/g standards. It has also been recognized lately as a possible candidate for use in 4^{th} generation mobile systems because of its many attractive benefits.

2.2.2 Orthogonality

In OFDM the sub-carrier pulse used for transmission is chosen to be rectangular. This has the advantage that the task of pulse forming and modulation can be performed by a simple IDFT which can be implemented very efficiently as an IFFT operation. As a result, the original pure line spectrum generates a *sinc* shaped sub-channel spectrum centered on each OFDM sub-carrier, as shown in Fig. 2.3.. In the receiver only an FFT operation is needed to reverse this operation.

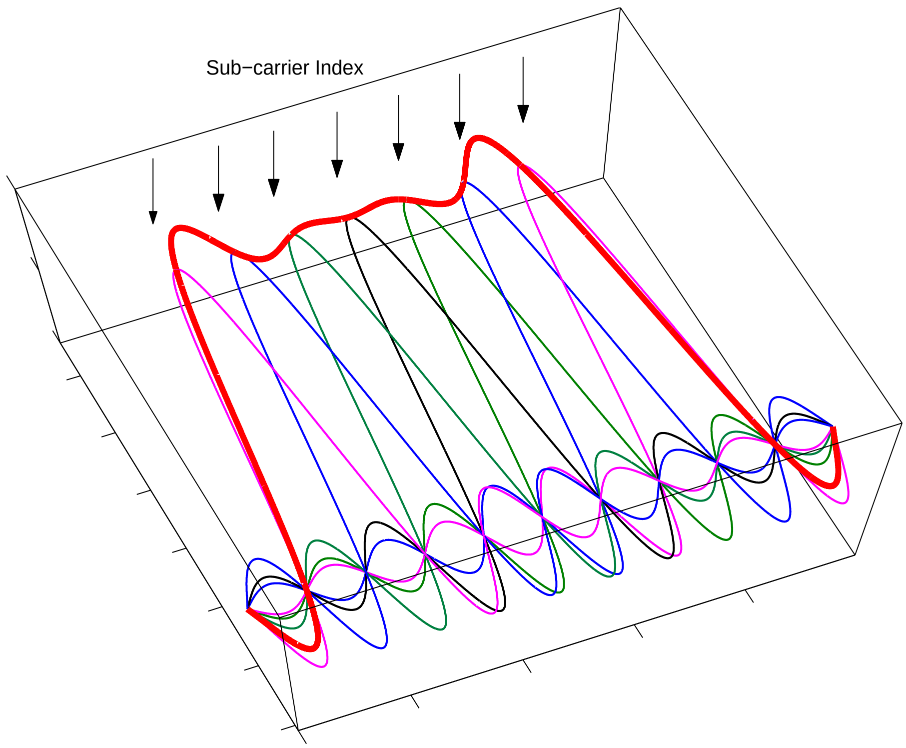

Figure 2.3. Spectrum of an OFDM Symbol Showing Orthogonal Sub-carriers.

As seen, the spectra of the sub-carriers are not completely separated, but overlap to some degree. The reason why the information transmitted over the carriers can still be separated is the so called orthogonality relation giving the method its name. By using an IFFT for modulation , the spacing of the sub-carriers is implicitly chosen in such a way that at the frequency where evaluation of the received signal is done (shown by arrows) is where all other signals are zero. To preserve perfect orthogonality, certain conditions need to be satisfied. The receiver and the transmitter must be perfectly synchronized, which means they both must assume exactly the same modulation frequency and the same time-scale for transmission. A more important condition is that there should absolutely be no multi-path, which is solved by cyclically extending the symbol by a guard interval (also called cyclic prefix), and the explanation of which is given in the following sections.

2.3 OFDM System Structure

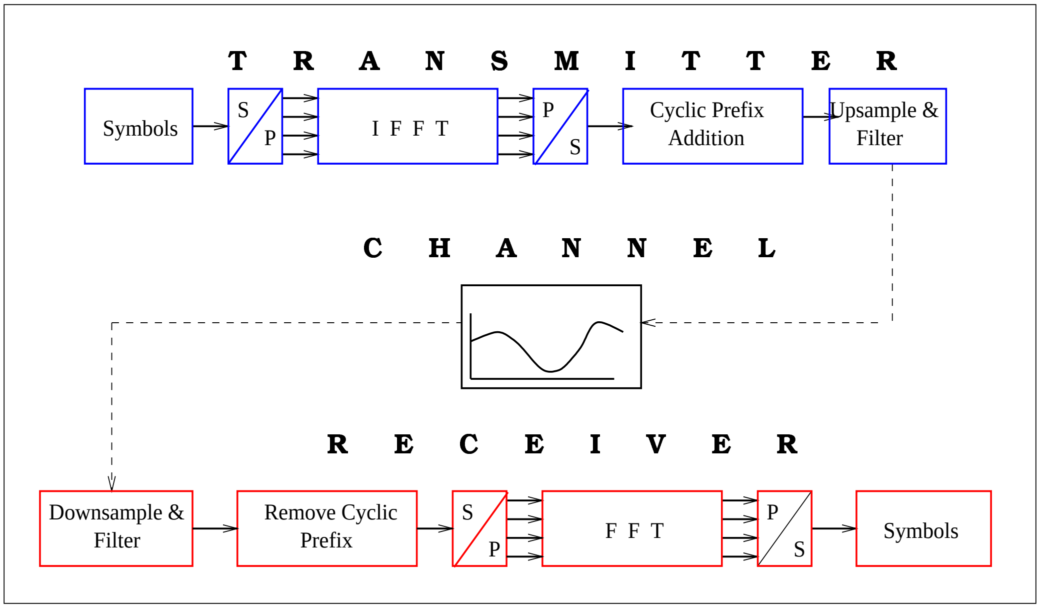

Figure 2.4. Block Diagram of a Typical OFDM Communication System.

A block diagram of a basic OFDM system is illustrated in Fig. 2.4.. Raw bits of data are coded and interleaved before modulation. In a multi-path environment, the occurrence of bit errors is usually restricted to only a few sub-carriers that undergo deep fade. To avoid this problem, channel coding can be used to correct errors to a certain level, depending on the code rate and type, and the channel. Interleaving is applied to randomize the occurrence of these bit errors. The coded and interleaved data is then mapped to the constellation points to obtain data symbols. Serial-to-parallel conversion of the symbols is done and IFFT is applied to these parallel blocks to obtain the time domain OFDM symbols. These samples are then cyclically extended as explained in later sections, converted to analog signals and up-converted to the RF frequencies using mixers. The signal is then amplified by a power amplifier (PA) and transmitted through antennas. At the receiver, the received signal is passed through a band-pass noise rejection filter and down-converted to baseband. After frequency and time synchronization, the cyclic prefix is removed and the signal is transformed to the frequency domain by using the FFT operation. The symbols are finally demodulated, deinterleaved and decoded to obtain the transmitted information bits. A detailed system explanation of the function of each block is presented in the following sections.

2.3.1 Serial to Parallel Conversion

The symbols are mapped onto N parallel sub-carriers, thus making the symbols on each sub-carrier N times longer than its serial counterpart. This reduces the effect of a time dispersive channel by making the symbol duration longer than the maximum excess delay of the channel ($\tau_m < T_s$) as shown in Fig. 2.5.. Each parallel stream of data is modulated onto a sub-carrier at a unique frequency and combined with the other sub-carriers to produce a serial stream of transmission data. Proper selection of transmission parameters can greatly reduce, if not eliminate ISI because the delay spread will be less than the symbol duration.

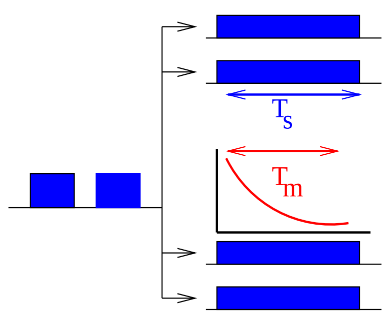

Figure 2.5. Serial to Parallel Conversion of Data Showing Increased Symbol Duration.

2.3.2 IFFT Operation

The idea to use the FFT to separate the sub-carriers in the frequency domain was proposed by Weinstein and Ebert [12], which greatly reduces the implementational complexity of OFDM and can be easily incorporated into practical systems. The parallel sub-carriers are seen as the spectrum of the signal to be transmitted as seen in Fig. 2.3., and hence the IFFT operation is performed to transform this spectrum to the time domain. The time domain symbols are obtained as,

$$\boxed{\begin{aligned} y(n) &= IFFT\{Y_k\} \\ &= \textstyle\sum_{k=0}^{N-1} Y_k e^{j2\pi nk/N} \qquad 0 \leq n \leq N-1 \,, \end{aligned}} \tag{2.2}$$

where Y_k is the symbol transmitted on the k^{th} sub-carrier and N is the total number of sub-carriers. At the receiver, the OFDM symbol can be demodulated using an FFT operation given by,

$$\boxed{\begin{aligned} Y_k &= FFT\{y(n)\} \\ &= \tfrac{1}{N} \textstyle\sum_{n=0}^{N-1} y(n) e^{-j2\pi nk/N} \end{aligned}} \tag{2.3}$$

2.3.3 Cyclic Prefix Insertion

The effect of time dispersive channels is suppressed, if not completely eliminated, by artificially extending the duration of the OFDM symbol. A copy of the last part of the OFDM symbol, termed as cyclic prefix shown in Fig. 2.6., is concatenated to the beginning of the transmitted symbol such that it is cyclically extended to be longer than the impulse response of the channel [13]. As seen from the figure, the distorted part of the signal will stay within the guard interval, provided the maximum excess delay is shorter than the length of the symbol. The guard interval thereby plays an important role in avoiding ISI.

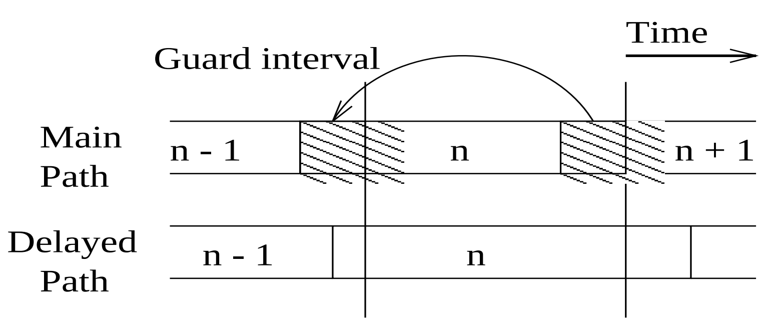

Figure 2.6. Illustration of the Guard Interval in an OFDM Symbol.

At the receiver, the samples of the cyclic extension are discarded. The disadvantage of the cyclic prefix is that it clearly reduces the the efficiency of the OFDM transmissions by a factor $N/(N+N_g)$, where N is the total number of sub-carriers and N_g is the length of the guard interval, or cyclic prefix. This is an acceptable trade-off when considering the advantages. Typically, a guard interval length of not more than 10% of the OFDM symbol duration is employed.

2.3.4 Windowing

A series of IFFTs that are concatenated to each other constitute an OFDM signal. At each symbol boundary, there is a signal discontinuity due to the difference between the end of one symbol and the start of another one. These very fast transitions at the boundaries increase the side-lobe power. In order to smooth the rapid transitions between different transmitted OFDM symbols, windowing is applied to each symbol.

2.3.5 Filtering

As observed from the block diagram, the filtering operation is performed both at the transmitter as well as the receiver. At the transmitter, filtering is used to limit the effect of spectral side lobes of the *sinc* shape of the OFDM sub-carriers. Digital filters are generally used due to their superior accuracy and cut off rate, as compared to their analog counterpart. The disadvantage of filtering is that it cuts of significant energy from the outer sub-carriers thereby resulting in their shape distortion and causing Inter-Carrier Interference (ICI). Some commonly used filters are rectangular pulse (*sinc* filter), root raised cosine filter, Chebyshev and Butterworth filters. At the receiver, a matched filter is used to reject the noise and adjacent channel interference.

2.4 Drawbacks of OFDM

1. Carrier Offset: OFDM is extremely sensitive to carrier frequency offset which manifests itself as a shift in the received spectrum of the signal in the frequency domain. In the presence of frequency offset, the sub-carriers are sampled at the wrong points which leads to spilling of energy between the sub-carriers causing deterioration in BER

performance. This effect is termed as ICI. Mismatch between the local oscillators at the transmitter and receiver and doppler spread are the factors contributing to ICI.

2. Peak-to-Average Power Ratio: OFDM signals sometimes exhibit a large instantaneous peak with respect to the average signal level *i.e.*, a large Peak-to-Average Power Ratio (PAPR). This occurs because the OFDM symbol is a superposition of a large number of signals on various sub-carriers, which may sometimes add constructively. This sometimes results in the signal going beyond the linear range of power amplifiers, and consequent clipping of the signal produces high out-of-band distortions and loss of orthogonality between the sub-carriers.

3. Sensitivity to Phase Noise: Phase noise in the oscillators of both the transmitter and receiver is another limiting factor in the performance of OFDM systems. It originates from oscillator inaccuracies and has the effect of additional phase and amplitude modulation of the received samples in baseband. The phase noise effect can be also visualized as an additional multiplicative effect of the channel, similar to slow or fast fading.

2.5 Applications

In the past, OFDM applications have been scarce because of the complexity involved in their implementation. Recent advances in DSPs and VLSI technology have enabled cost-effective and practical implementation of the DFT/IDFT via the FFT/IFFT operation on a single chip, and have spurred an array of applications such as their adoption as the next European digital audio broadcasting standard (DAB) [1] and for Terrestrial Video Broadcasting (DVB-T) systems. OFDM is also employed for fixed-wire applications such as Asynchronous Digital Subscriber Line (ADSL) [2], and High bit-rate Digital Subscriber Line Systems (HDSL) [3]. Most recently, the standardized High PERformance Local Area Network standard known as HIPERLAN/2 in Europe and the IEEE 802.11a/g, that were both designed for indoor wireless networking also incorporate OFDM. Another standard termed 802.11h is being devised, which is an extension of the 802.11a with transmission power control and dynamic frequency selection. In 1993, the Multi-carrier CDMA (MC-

CDMA) system, which is a combination of both OFDM and CDMA was proposed. This system has many interesting features that has garnered much attention for possible use in 4^{th} generation mobile systems. Most recently, the 802.16a standard amended by the IEEE as a WMAN networking protocol, uses OFDM in the physical layer specifications. It is aimed at enabling a wireless alternative for cable, DSL and T1 services for last-mile broadband access .

The future of wireless communications is aimed at achieving ubiquitous and high-speed access to multimedia information possible. Future users will demand multimedia communication services, where multimedia points out a communication with multiple ways of presenting the information *i.e.* as a combination of text, data, graphics, animation, images, sound, speech and still or moving video. In the search for a suitable technology that would enable such a vision, OFDM based systems promise to be an excellent choice.

CHAPTER 3

ADAPTATION IN OFDM SYSTEMS

In this chapter, the concept of adaptation in reference to OFDM systems is introduced. Adaptation is the process whereby the transmission parameters are modified on a burst-by-burst basis according to the perceived channel conditions. Several parameters such as modulation mode, code length, guard period length, etc. can be dynamically modified based on channel integrity in order to maximize throughput and system resource utilization. Adaptation of these parameters involves three enabling techniques namely channel quality estimation, transmission parameter modification and their proper detection by the receiver. First, the motivation behind adaptive modulation and its evolution is introduced. This is followed by a brief summary of the enabling techniques and their respective algorithms proposed in literature. Finally, alternative schemes proposed in literature in reference to OFDM systems, to combat the frequency selectivity of the channel, are presented.

3.1 Motivation

In recent years, the concept of mobile multimedia wireless transceivers has gained immense popularity. Numerous applications and services have emerged, the sustenance of which requires the transceiver to constantly adapt itself to changing channel conditions to maximize the utility of available resources. The fundamental throughput and capacity limiting factor of any wireless system is attributed to time and frequency domain channel fading as illustrated in Fig. 3.1., which was generated with the aid of the FFT of the Rayleigh faded channel impulse response. It shows how the instantaneous channel SNR is a function of both time and frequency.

As can be seen, the bit rate of a fixed-mode transceiver would fluctuate as a function of time. The SNR fluctuations observed in time and frequency is an indication that a fixed-

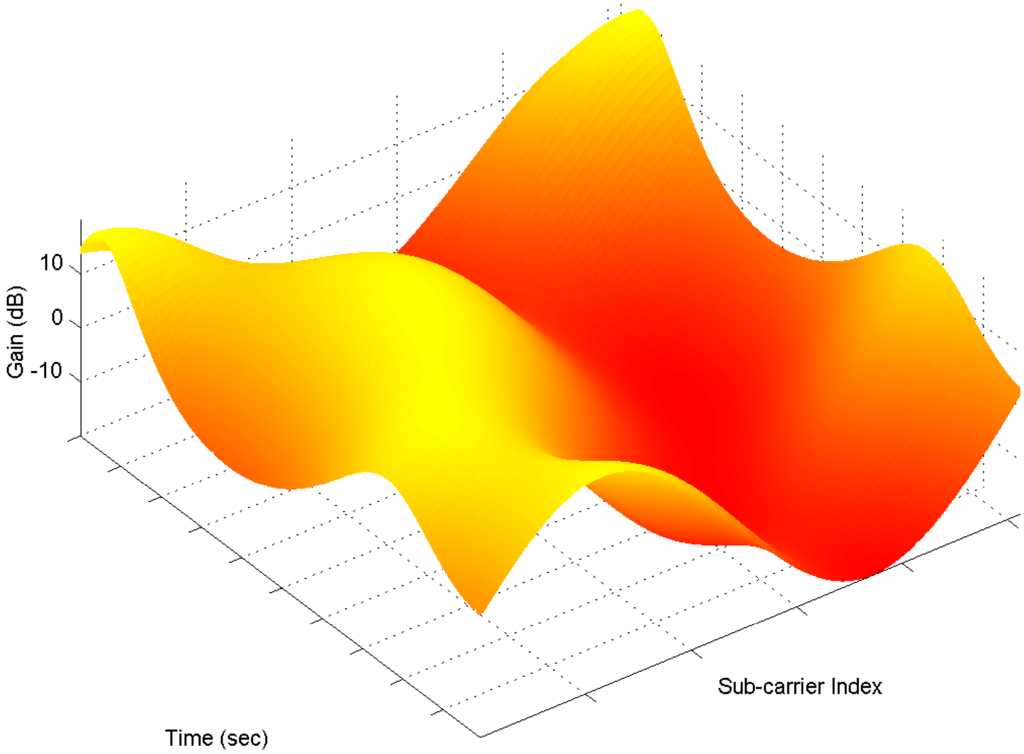

Figure 3.1. A Typical Two Dimensional Channel Frequency Response.

mode transceiver cannot guarantee satisfactory performance and maximize system resource utilization in these scenarios, while still providing acceptable complexity requirements. This is the key motivating factor that has gathered considerable research interest in the area of variable-mode or adaptive transceivers. The principle behind adaptation is to devise an efficient approach to counteract the detrimental effects of the frequency selective fading channel, by varying the transmission parameters based on instantaneous channel quality measurements [14]. This would result in higher throughput communication and increase system resource utilization over hostile channel conditions, as compared to a conventional fixed-mode transceiver. As can be observed from Fig. 3.1., the bit error probability of different sub-carriers of the OFDM symbol is restricted to certain severely degraded sub-carriers, while the rest of the sub-carriers often suffer from little or no fading. In other words, frequency selective fading affects certain sub-carriers more severely than others. Adaptive modulation functions in such a way that the transmission parameters are adaptively changed based on the estimated channel transfer function. The faded sub-carriers use a more robust modulation mode, e.g. BPSK, or if needed, are excluded from any data transmission. This potential loss of throughput can be compensated by employing higher order modulation schemes on the sub-carriers exhibiting high SNR. The net effect of adapting the modulation scheme is that the overall BER of the system can be improved, and the throughput can be increased simultaneously. Adaptive OFDM systems have the advantage of tracking the signal quality both in the time and frequency domain. Often, sub-carriers are grouped together and adaptation is performed on an entire sub-carrier group to reduce complexity. This is illustrated in Fig. 3.2., which shows the different modulation schemes used within different sub-carrier groups based on instantaneous channel conditions.

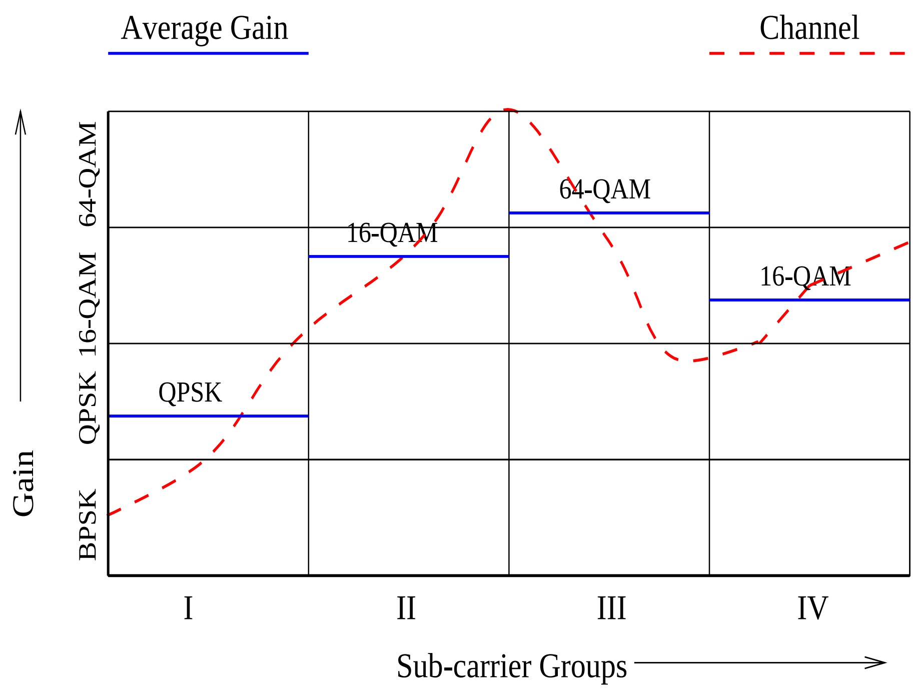

Figure 3.2. Illustration of Sub-band Adaptive Modulation.

The performance of adaptive transceivers can be characterized by numerous indicators. Adaptive modems outperform their fixed-mode counterparts in frequency selective fading channels since for a given number of bits per symbol (BPS), their average BER will be lower. From a different standpoint, given a certain BER, their BPS will always be higher than conventional fixed-mode transceivers. When employing these adaptation techniques in the frequency domain, such as in conjunction with OFDM, attractive system design trade-offs and exceptional system performance can be gained. On the same note, the performance of adaptive systems requires that the transceiver react rapidly to changes in channel conditions. The steps entailing effective performance of adaptive systems are efficient channel quality estimation, optimal parameter selection and detection of the modified parameters at the receiver.

Despite these numerous advantages, extensive research in this field needs to be done in order to solve a plethora of other problems associated with adaptive modulation. The algorithms used in the estimation and detection of the modified parameters must be optimally tailored to the adaptive OFDM system structure, failing which it will lead to the selection of the wrong set of parameters thereby degrading system performance. Signal processing techniques and other enabling methods need to be improved to further evolve the aforementioned enabling techniques to enhance the performance of adaptive OFDM systems.

3.2 Literature Overview

As mentioned, the aim of adaptive modulation is to maximize the available channel capacity of frequency selective fading channels by changing the transmission parameters based on the instantaneous channel conditions. From a historical standpoint, interest in adapting the modulation scheme and transmission parameters began with Hayes [14], who described the adaptation of signal amplitude based on observed channel conditions. The performance of such adaptation schemes can be improved by employing better channel estimation and prediction methods [15]. To counteract the frequency selective nature of the channel, Hayes [14] proposed the usage of transmission power adaptation, which has a disadvantage of producing co-channel interference. Others, such as Cavers [16], suggested the use of variable symbol duration. The drawback of increasing symbol duration is bandwidth wastage, and is not considered an optimal solution for contemporary systems. These disadvantages were addressed by Steele and Webb [17], who proposed Adaptive Quadrature Amplitude Modulation (AQAM) scheme variation for fading narrow-band channels, and suggested that AQAM exhibited promising advantages in terms of spectral efficiency when compared to fixed modulation schemes.

The introduction of Pilot Symbol Assisted Modulation (PSAM) led to the usage of square-shaped AQAM constellations rather then star-shaped ones in an attempt to counteract fading. With the analysis of the channel capacity of Rayleigh fading channels [18], Goldsmith [19] and Alouini [20] proved that a combination of variable power and variable rate adaptive techniques are more desirable in terms of maximal channel utilization. Torrence and Hanzo [21] proposed a set of modulation mode switching levels which were specifically designed for high throughput, while simultaneously maintaining a target BER over a narrow-band Rayleigh fading channel. These results were further extended and improved by Choi and Hanzo [22], who went on to formulate a set of SNR dependant switching thresholds to maximize a target BER based throughput.

The aforementioned references point to the fact that AQAM has significant advantages in terms of spectral efficiency and BER performance. Following these developments, adaptive modulation was also studied in conjunction with channel coding and power control techniques [23]. Sampei and Morinaga [24] mention symbol rate adaptive modulation where the symbol rate or the modulation modes was varied by using $\frac{1}{8}$-rate 16-QAM, $\frac{1}{4}$-rate 16-QAM and $\frac{1}{2}$-rate 16-QAM as well as full rate 16-QAM. The switching criterion used was the estimated channel SNR and delay spread. Channel coding rate variation was introduced by Matsuoka [25] where the transmitted burst utilized an outer Reed-Solomon Code and an inner convolutional code to achieve good performance. The code rate was varied based on a target BER requirement, similar to adaptive modulation mode switching. The performance of adaptive channel coding in reference to adaptive modulation in a narrow-band environment was studied by Chua and Goldsmith [23], where lattice and trellis codes were used. Goeckel [26], in his contribution, characterized the system performance by using outdated fading estimates rather than perfect estimates. Power control in conjunction with a predistortion type non-linear amplifier compensator in reference to adaptive modulation was studied in [27].

The associated principles can also be applied in the context of multi-carrier OFDM systems [28]. This principle was first presented by Kalet [29] and further developed by Chow, Coiffi and Bingham [3] and also by Czylwik [30]. Adaptation of transmission parameters is done on a timeslot-by-timeslot basis, based on the transmitter's perception of the channel.

Therefore, a slowly varying channel is assumed because the channel characteristics are extrapolated based on previous timeslot observations. Adaptation on a timeslot-by-timeslot basis has been investigated in [31] and has found to improve the BER performance for duplex reciprocal slowly varying channels.

The performance of adaptive modulation systems implicitly ascribes to the fact that the algorithms used to determine channel quality must perform effectively for proper parameter selection. Other methods such as parameter adaptation, and detection techniques must work on par with one another to reap maximal benefit and optimal performance.

3.3 Enabling Techniques

Optimal performance of adaptive OFDM systems is dictated by the performance of the enabling techniques that entail its operation. If these methods that support adaptation do not perform effectively, it will lead to a selection of the wrong set of transmission parameters thereby degrading system performance. These techniques are briefly described in the following sections.

3.3.1 Parameter Estimation

The fourier transform of the OFDM symbols yields the received samples in frequency domain, which can be shown as,

$$\boxed{\begin{aligned} Y_k &= FFT\{y_n\} \\ Y_k &= S_k H_k + Z_k \end{aligned}} \tag{3.1}$$

where H_k and Z_k are the channel frequency response and the FFT of the unwanted interference respectively. Reliable methods to estimate the instantaneous channel transfer function (H_k) and the frequency domain variation of noise power (Z_k) are needed in order to modify transmission parameters such as modulation mode, coding rate etc., according to the instantaneous channel conditions. It should be noted that adaptive OFDM systems perform well only in slowly varying channel conditions. This is because prior knowledge of the channel can be gained only by extrapolating previously acquired channel information.

Several methods can be used to estimate the quality of the channel for OFDM based systems. The channel transfer function can be estimated using pilot tones which are inserted into all the OFDM sub-carriers at a specific period. A comparative study of various pilot based channel estimation and interpolation techniques can be found in [32, 33, 34]. More accurate measures of the channel transfer function can be obtained by means of decision directed or time domain training sequences based methods. The estimated channel transfer function does not take into account factors such as co-channel interference. Other channel quality measures are based on the error correction decoder's soft output information or by means of decision feedback local SNR estimation. Noise power estimation has been dealt with in a variety of ways in literature. Some consider SNR measurement as an indication of long-term fading statistics due to shadowing and log-normal fading. This long-term SNR estimate is often calculated using regularly transmitted pilot (or training) sequences. Instead of using pilot sequences, the data symbols can also be used for this purpose. For example, [35] uses SNR information as a channel quality indicator for rate adaptation and exploits the cumulative Euclidean metric corresponding to the decoded trellis path for channel quality information. Several other techniques which can be found in [36] and reference listed therein.

3.3.2 Adaptation of Transmission Parameters

Based on the observed channel, the transmission parameters such as modulation mode, etc., are varied to achieve a range of application dependant trade-offs between received data integrity and throughput. Deeply faded sub-carriers can also be excluded from data transmission and used for other purposes, such as PAPR reduction. A general parameter selection rule for an M configurable parameter adaptive modulation system which would adjust the parameter m, where $m \in (1, 2, \ldots M)$, based on channel quality indicator ϑ is given by,

$$\boxed{Modify\ parameter\ m\ if\ s_m \leq \vartheta < s_{m+1}} \tag{3.2}$$

where the switching level $s_m \in (s_m \mid m = 1, 2, 3 \ldots M)$. Several criteria to modify transmission parameters can be used depending on the application. The most commonly used criteria are a target BER and throughput maximization. Optimal modulation switching levels, for example, to achieve a target BER in a flat-fading Rayleigh channel have been investigated by Torrence and Hanzo [21] with regard to OFDM systems. The criterion used here was the instantaneous received power, which was estimated by exploiting the reciprocal nature of the channel in a time division duplexed (TDD) environment. This estimate is then used to select a suitable modulation scheme based on a set of switching levels.

The channel coding parameters such as coding rate, variation of block lengths for block codes, adaptive interleaving and puncturing for turbo and convolutional codes can also be varied based on perceived channel conditions as investigated in [28].

Spectral pre-equalization is another method to improve the BER performance of the system [37]. It is done by completely inverting the frequency domain channel transfer function at the transmitter. The channel inversion is done at the transmitter (base station) because it allows easy integration of low-cost mobile terminals and superior down-link performance. Exact counteraction of the frequency domain channel transfer function is seldom possible due to output power restrictions, and so slight variations of the spectral pre-equalization in conjunction with adaptive modulation are formulated [38]. They involve methods to block some severely faded sub-carriers from transmitting data to limit the output power of the transmitter.

3.3.3 Signaling/Blind Detection of Parameters

Unlike fixed-mode transceivers, adaptive systems constantly change their transmission parameters based on observed channel conditions. The modified transmission parameters across the OFDM sub-carriers need to be communicated effectively to the receiver for proper demodulation and decoding of the received OFDM symbols. This can be achieved either by explicit signaling on pre-specified sub-carriers, or blind detection methods [38, 39]. To avoid excessive overhead information, efficient signaling or blind detection schemes have to be employed in adaptive OFDM systems. A simple form of signaling the modulation mode to the receiver is by means of an M-PSK symbol, M being the total number of possible modes used by the system. Each constellation point of the M-PSK symbol denotes a specific modulation scheme used in a specific sub-band as shown in in Fig. 3.3..

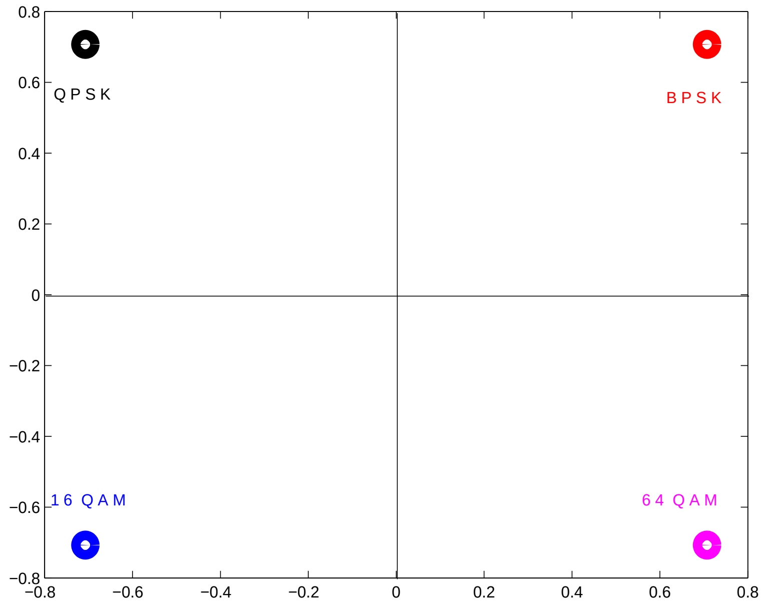

Figure 3.3. Using a QPSK Symbol to Signal Modulation Modes.

These M-PSK may be interleaved across the OFDM sub-carrier index in order to obtain frequency diversity. Early research in reference to signaling of required modulation mode was restricted to narrow-band channels. Efficient modulation control parameter transmission was proposed by [40], where the parameters are embedded within the OFDM symbol using Walsh codes and subsequent decoding at the receiver was done by maximum likelihood detection. The disadvantage of using these signaling symbols is the resulting loss in throughput. An alternative method is to estimate the modulation mode from the received noisy data symbols. Two methods in this regard have been mentioned in [5], which are based on SNR estimation and error correction coding. Detection based on SNR estimation is based on finding and minimizing the difference between the noisy received sample in frequency domain, and the best hypothesis of the noiseless received sample. Assuming that the channel state information on each carrier is known, SNR can be estimated and the modulation mode used can be found out. Detection Error Rates (DER) of less than 10^{-3} are observed for channel SNR values of 15-18 dB for four modulation options namely BPSK, QPSK, 16-QAM and 64-QAM.

Blind modulation mode detection is an attractive alternative to signaling mainly because it increases the overall throughput and data capacity of adaptive OFDM systems, and further research needs to be focussed on this enabling technique.

3.4 Spectral Pre-equalization

In conventional pilot based frequency domain equalization, the estimated channel transfer function is used to nullify the amplitude and phase distortion effects of the channel. The received OFDM symbol consists of the faded transmitted symbol and unwanted additive noise. The estimated symbol $\hat{Y}_k$ is obtained by,

$$\hat{Y}_k = \frac{Y_k}{\hat{H}_k} = \frac{S_k \cdot H_k}{\hat{H}_k} + \frac{Z_k}{\hat{H}_k} \tag{3.3}$$

where the $\hat{H}_k$ estimated transfer function.

A disadvantage associated with this pilot based method is that even if $\hat{H}_k$ is accurate, along with de-fading the symbol there is amplification of the unwanted noise by the same amount. This fails to improve the SNR of the received symbol. An interesting alternative to combat the frequency selective nature of the Rayleigh fading channel is introduced in [41]. Spectral pre-equalization requires the estimate of the channel, and it uses this information to scale the OFDM symbols so as to null the effect of the channel, and reduce it to a flat fading one. Pre-equalization is applied to the OFDM symbols prior to transmission. It should be noted that, similar to adaptive modulation, this method can only be applied in a TDD link.

If the channel transfer function is available at the transmitter end, then depending on the fading level of the sub-carriers (or band of sub-carriers), the symbols are scaled by a pre-equalization function. This is calculated by the inverse of the frequency domain channel transfer function. The goal is to nullify the effect of the channel before transmission so that no equalizer is needed at the receiver. Mathematically, pre-equalization is shown as Eq. (3.4).

$$\boxed{R_k = S_k H_k \cdot P_k + N_k} \tag{3.4}$$

where P_k is termed as the pre-equalization function. The exact operation is schematically shown in Fig. 3.4..

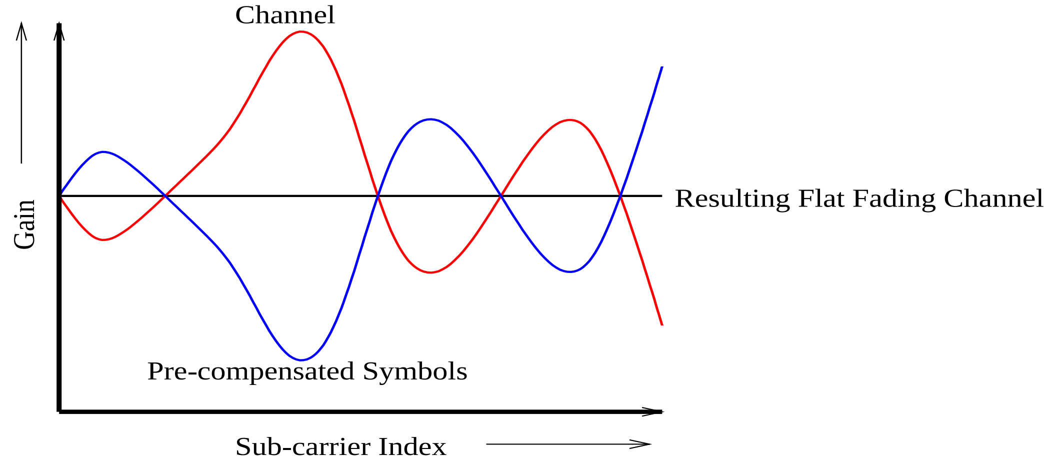

Figure 3.4. Pre-equalization to Counteract the Effect of a Fading Channel.

Since there is no equalization performed at the receiver, there is no amplification of the unwanted noise, and the SNR of the received symbol is improved. Two methods by which pre-equalization can be used to yield better results are mentioned in [37]. The motivation for this alternate approach is that sometimes direct channel inversion has the effect of excessive output power, which would place strict restrictions on the power amplifier requirements. A brief explanation of these methods is given in the following sections.

3.4.1 Pre-equalization with Sub-band Blocking

In typical Rayleigh fading channels, it is not often possible to directly reverse the effect of the channel. If direct inversion is done, it will lead to power fluctuations whose severity depends on the degree of channel variation. If deep fades occur in any particular set of sub-carriers, it is more practical not to send data on these sub-carriers because no matter what modulation scheme used, the signal level may be lower than the noise floor and therefore cannot be detected. Rather, the unused power (the power otherwise used to send data on the faded sub-carriers) can be used to increase the strength of the remaining data carrying sub-carriers which would increase the SNR at the receiver. This is the concept behind pre-equalization with sub-band blocking. The blocked sub-band is the set of sub-carriers that are in deep fade. This scheme is possible only in a duplex TDD scenario because the transmitter needs an estimate of the current frequency domain channel transfer function. This is obtained by the received signal in the reverse link. The information on whether or not a sub-band is used for transmission has to be signaled to the receiver.

3.4.2 Adaptive Modulation with Spectral Pre-equalization

Pre-equalization with sub-band blocking has a disadvantage, *i.e.* it does not attempt to negate the channel's path loss. It only inverts the channel's transfer function so that it resembles a Gaussian flat-fading channel and has only been put forward as a means to limit the transmitter's output power to maintain a nearly constant SNR over the entire sub-carrier index. Adaptive modulation with pre-equalization is used in situations that do not demand a constant throughput, but require a particular BER at a given power level. Therefore, it is not entirely a pre-equalization algorithm. By clubbing both adaptive modulation and spectral pre-equalization, the transmitter would react to the time and frequency variations of the channel to tune to adaptive modem appropriately. Also, the added advantage similar to that in the previous method is that the energy not used in transmission can be redirected to other data bearing sub-carriers. The receiver would have to anticipate transmission in different power levels for various modulation schemes employed so that error-free demodulation can be achieved. If constant throughput is not a criterion, then a fixed BER scheme with error correction coding can be used to maximize throughput. The algorithm would require the estimate of the noise floor at the receiver and the channel transfer function. It is on this basis that the necessary amplitude required on each data symbol is selected for a given sub-carrier.

CHAPTER 4

NOISE POWER AND SIGNAL-TO-NOISE RATIO ESTIMATION

Noise Power and SNR estimation are important parameters in digital communications since they serve as a standard measure of signal quality. The received signal is usually a combination of the faded transmitted signal and unwanted additive noise. Conventional algorithms assume that the statistics of the unwanted additive noise remain constant and hence average the instantaneous noise samples over all the OFDM sub-carriers. In reality, this noise is a combination of white and correlated or colored noise. Colored noise is characterized by a non-constant power spectrum. This chapter introduces a windowing technique to estimate noise power and takes into account the variation of the noise statistics over the OFDM sub-carrier index.[1] The proposed method provides many local estimates of noise power, allowing tracking of the variation of the noise statistics across OFDM sub-carriers. This is particularly useful in sub-band adaptive OFDM systems. The mean-squared-error (MSE) condition for choosing the optimal window size is derived, and a sub-optimal method for practical implementation in sub-band adaptive modulation systems is presented. Computer simulations show that the proposed method tracks the local statistics of the noise effectively in the presence of colored noise in addition to white Gaussian noise, and reduces to the conventional methods when it is white thereby exhibiting robustness.

4.1 Introduction

SNR is broadly defined as the ratio of the desired signal power to the unwanted noise power. It has long been accepted as a standard measure of signal quality in noise corrupted environments. Noise originates from many sources including the external environment, non-linear hardware components, etc. Efficient system adaptation requires the estimate of SNR

[1]This work is partly published in [4]. An extended version with mathematical proof has been submitted to RAWCON 2004.

in order to modify the transmission parameters to maximize efficiency of system resources. Poor channel conditions, reflected by low SNR values, requires that the transmitter modify transmission parameters such as coding rate, modulation mode etc., to satisfy certain application dependant constraints such as constant BER or throughput. Adaptivity requires an efficient and reliable real-time SNR estimator for continual channel quality monitoring and compensation to maximize resource utilization and QoS.

The performance of the SNR estimator is of crucial importance in the overall performance of the adaptive OFDM system, which employs different modulation modes on groups of sub-carriers based on the observed SNR in frequency domain. There are many other applications that can exploit SNR information, like channel estimation through interpolation and optimal soft information generation for high performance decoding algorithms [42].

4.2 Literature Overview

SNR measurement in previous estimation methods is considered an indication of long-term fading statistics due to shadowing and log-normal fading. This long-term SNR estimate is often calculated using regularly transmitted training (or pilot) sequences. Data symbols can also be used for this purpose. For example, Balachandran [35] uses the SNR information as a channel quality indicator for rate adaptation and exploits the cumulative Euclidean metric corresponding to the decoded trellis path for channel quality information. The early work of Benedict and Soong [43] from 1967 involves computation of SNR from higher-order averages of the envelope of a modulated signal. More than two decades later Shah and Hinedi [44] proposed to take advantage of the correlation properties of the signal and noise in narrow-band channels. Jacobsmeyer [45] describes another method for channel quality measurement and proposes the use of the difference between the maximum likelihood decoder metrics for the best path and the second best path [45]. However, this approach does not provide any information about the strength of the interferer or the desired signal. There are several other SNR measurement techniques which can be found in [36] and reference listed therein.

In many SNR estimation techniques, the interfering noise is assumed to be white and Gaussian distributed. However, in practical wireless communication systems, noise is often caused by a strong interferer which is colored in nature. Color of the noise is characterized by the variation of its power spectral density. It is due to this uneven variation that some parts of the OFDM spectrum are affected by the interferer more than the other parts. Fig. 4.1. shows the OFDM frequency spectrum and both types of noise over this spectrum, namely colored and white.

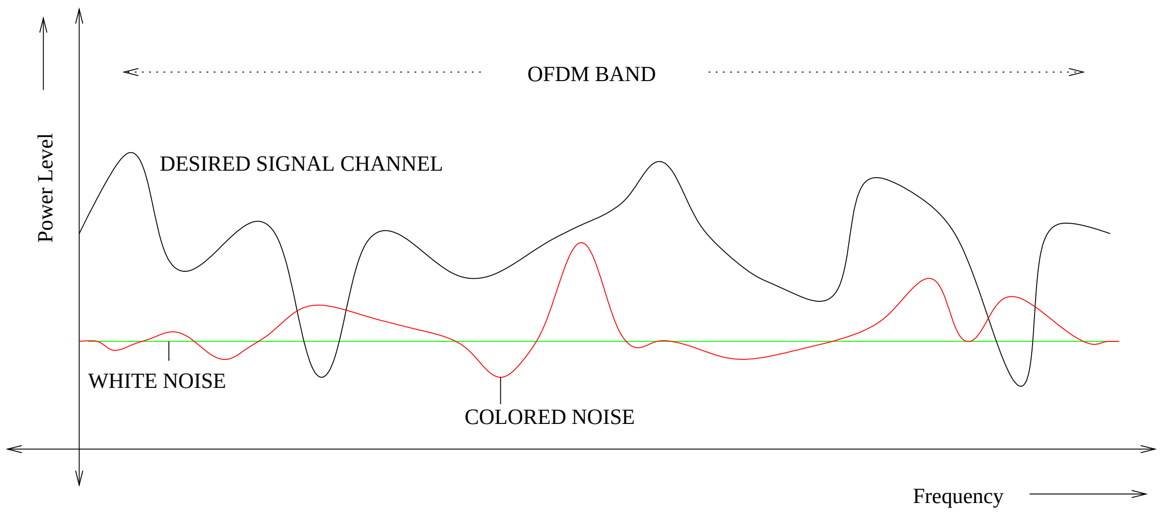

Figure 4.1. Representation of OFDM Frequency Channel Response and Noise Spectrum. Spectrum for White and Colored Noise.

In many new generation wireless communication systems, coherent detection is employed, which requires estimation of channel parameters. These channel parameter estimates can also be used to calculate signal power. Therefore, SNR estimation introduces only an additional estimation of noise power. The focus is more on estimation of noise power, and it is assumed that the signal power can be estimated from the channel estimates. Most commonly used approach for noise power estimation in OFDM systems is based on finding the difference between the noisy received sample in frequency domain, and the best hypothesis of the noiseless received sample [5, 46]. Calculation of the received sample hypothesis requires channel state information for each carrier. As mentioned, the previous approaches estimate long term noise power values, assuming that the noise is white and Gaussian distributed. In order to get the long term estimates, the instantaneous SNR estimates are averaged over the whole OFDM transmission spectrum by taking the mean of all the estimates over all the sub-carriers.

4.3 System Model

OFDM converts serial data stream into N parallel blocks and modulates these blocks using the IFFT operation. Time domain samples of an OFDM symbol can be obtained from frequency domain symbols by the IFFT operation given in Eq. (2.2). After the addition of cyclic prefix and D/A conversion, the signal is passed through the mobile radio channel. Assuming a wide-sense stationary and uncorrelated scattering (WSSUS) channel, the channel $H(f,t)$ can be characterized for all time and all frequencies by the two-dimensional spaced-frequency, spaced-time correlation function,

$$\boxed{\phi(\Delta f, \Delta t) = E\{H^*(f,t)H(f+\Delta f, t+\Delta t)\}} \tag{4.1}$$

We assume the channel to be constant over an OFDM symbol, but time-varying across OFDM symbols, which is a reasonable assumption for low and medium mobility.

At the receiver, the signal is received along with the interfering noise. The simplified received baseband model of the samples after synchronization, down-sampling, and removing the cyclic prefix can be written as,

$$\boxed{y(n) = \sum_{l=0}^{L-1} x(n-l)h(l) + z(n)} \tag{4.2}$$

where L is the number of channel taps, $z(n)$ is the noise sample which is combination of white Gaussian noise and colored interference, and the time domain channel impulse response (CIR), $h(l)$, over an OFDM symbol is represented as time-invariant linear filter. After taking the FFT of the OFDM symbols, the received samples in frequency domain can be represented as

$$\boxed{Y_k \quad = \quad S_k H_k + Z_k \ ,} \tag{4.3}$$

where H_k and Z_k are FFT of $h(l)$ and $z(n)$, respectively.

In practical wireless communication systems, often the received signal is impaired by dominant interference sources which are not white and Gaussian distributed. For example, in cellular systems, the dominant interference source can be a co-channel or an adjacent channel interferer. In WLAN systems, this can be a Bluetooth interference (like in 802.11g), or due to any other colored dominant noise source (like baby monitors, cordless phones, microwaves, etc.). This motivates the adoption of an alternative approach to noise power estimation to effectively track the varying noise statistics across the OFDM sub-carriers.

4.4 Noise Power Estimation

4.4.1 Advantage of Windowing

Consider $z(n)$ to be the time domain colored noise plus white interference signal. The Fourier transform of $z(n)$ would give the frequency domain version, *i.e.* Z_k. In conventional noise power estimation algorithms [46], the absolute squares of Z_k, which are the instantaneous noise samples, are averaged over all OFDM sub-carriers, thereby providing a single averaged noise power estimate. The conventional algorithms assume the noise to be white Gaussian distribution and estimates the averaged noise power for all the OFDM sub-carriers. Therefore, these approaches do not provide any information about the variation of noise within the transmission bandwidth, which is crucial for sub-band adaptive modulation systems.

In the proposed sub-optimal windowing approach, the whole band (*i.e.* the total number of sub-carriers) is divided into sub-bands (*i.e.* to a set of sub-carriers), each of size W. If the number of sub-carriers in each sub-band is N, then the number of sub-bands will be N/W. Then, the absolute square of the instantaneous noise estimates in each sub-band are averaged,

$$\hat{E}_s = \frac{1}{W}\sum_{l=1}^{W} E_l \ , \qquad\qquad 1 \leq s \leq N/W \tag{4.4}$$

where $\hat{E}_s$ is the estimated power in the s^{th} sub-band. The size of W (window size) depends on the color of the noise. If the noise is completely white, then it is desired that averaging be done across all the available OFDM sub-carriers, *i.e.* to have W equal to N. Note that if the noise is white and Gaussian distributed, $\hat{E}_s$ has a chi-square distribution with W degrees of freedom. The variance of E_s is the MSE of the noise power estimator. Therefore, increasing the number of samples over which averaging is done yields a lower MSE in the case of white noise, but the same does not apply for colored noise. The averaged noise estimates over each sub-band are further averaged across several OFDM symbols (averaging in time). Therefore, instead of having one noise power estimate for all the sub-carriers, we obtain N/W estimates, each representing the noise power estimate over each sub-band. The averaging window in time depends on the variation of the sub-band noise power in time. If the noise statistics vary rapidly, then it is desired to choose smaller window size in order to be able to track the variations of the noise power. On the other hand, if the noise statistics change slowly, it is desired to have a larger window size to allow better averaging of the sub-band noise values.

4.4.2 Preliminary Observations

The preliminary performance results, which demonstrate the advantage of using the windowing technique, are presented. An OFDM system with 64 sub-carriers is considered. For colored noise, a correlated co-channel interference source is used. The channels for the desired and interfering signals are uncorrelated, and varying in frequency (frequency selective) and in time (time selective). The channel taps are obtained using the modified Jake's fading model [47]. Frequency variation depends on the root-mean-squared (rms) delay spread of the channels, and time variation depends on the Doppler spread. Carrier frequency of 5.2 GHz, and OFDM symbol duration of 4 μsec are considered in the simulations. Additive white noise is included apart from colored noise, and performance evaluations are made for a white and color noise dominated environments, which are created by varying their respective ratio.

Fig. 4.2. compares the conventional and proposed noise power estimates over one realization of the channel. The average signal power is normalized to 0 dB and signal-to-

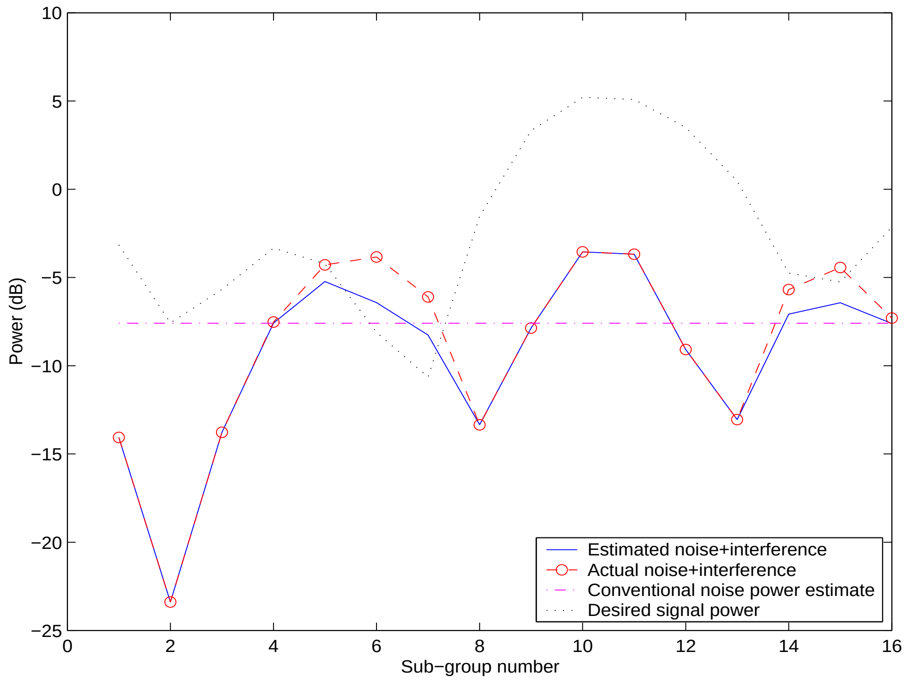

Figure 4.2. Comparison of Conventional and Proposed Noise Power Estimation Algorithm. A Realization of Signal and Noise Powers over the Transmission Band.

interference ratio of 7 dB is used in this figure, *i.e.* the average noise power is -7 dB. As described before, the conventional noise power estimate measures only a single value over the whole frequency band. This value is the averaged noise power over the whole band. As can be seen, the conventional estimate works well in measuring the average noise power. The proposed scheme measures both local and global noise power estimates. The total band is divided into 16 groups of sub-bands and in each sub-group, there are 4 sub-carriers. The instantaneous noise power values are averaged over each sub-group. Then, these averaged values over each sub-group are further averaged over 50 OFDM symbols (averaging in time domain). The figure shows the averaged noise power estimates over each sub-group. For reference purpose, the actual noise power values in each sub-group are also given. As can be seen, the proposed algorithm estimates the local noise power very well. In the figure the desired signal power over each sub-group is also given. From these, SNR values over each sub-group can be calculated easily. Notice that the noise power estimate performance depends on the SNR value over each sub-group. When the SNR value is very low, the estimates deviate from the actual noise power value due to incorrect decisions. As described before, the estimates can be improved further by using decoded decisions.

Fig. 4.3. shows the MSE performance of the proposed and conventional algorithms in colored noise dominated environments. The interference limited scenario as mentioned above is considered with different interference power levels. The MSE between the actual noise and estimated noise values in each sub-group are calculated and averaged. As can be

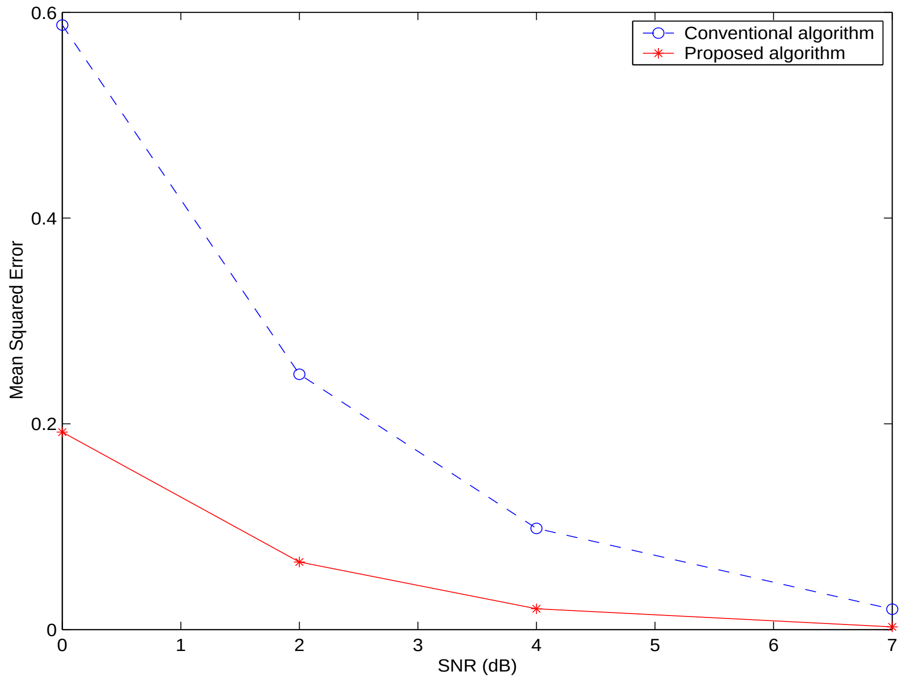

Figure 4.3. Mean-Squared-Error Performance of Conventional and Proposed Algorithms in Colored Noise.

seen the proposed algorithm performs much better than conventional noise power estimation in terms of finding the local noise power.

Figure 4.4. shows the MSE performance in white noise. This figure shows the robustness of the proposed algorithm when the noise is not colored. As can be seen, the proposed algorithm works as well as the conventional algorithm which is specifically designed for white noise assumption. Even if the noise is white, the proposed algorithm does not lose performance against conventional scheme. On the other hand, as explained above, if the noise is colored, the proposed algorithm outperforms conventional noise power estimation algorithm.

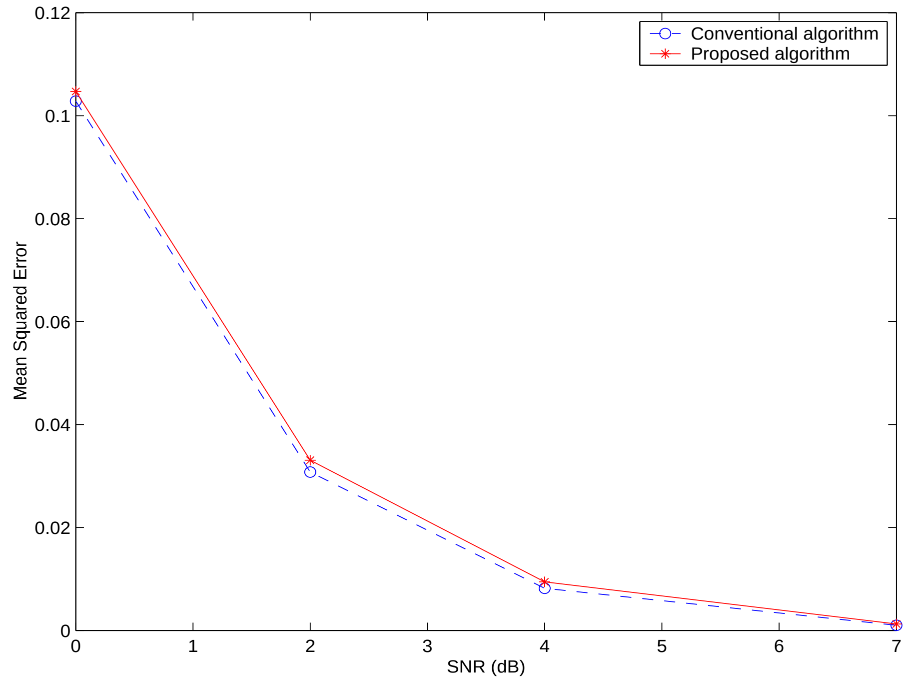

Figure 4.4. Mean-Squared-Error Performance of Conventional and Proposed Algorithms in White Noise.

4.4.3 Ideal Window Size

Initial results proved that the windowing technique tracks the varying noise statistics more effectively than conventional methods in the absence of colored noise. These results were presented for a fixed window size irrespective of the correlation characteristics of the interfering noise or variance of the white noise. The size of the window would depend on the correlation of the power spectral density of the colored noise and the additive white noise variance. It is known that when the noise is completely white, the largest window size would yield the lowest error between the estimated and true noise power. On the other hand, if the noise is completely colored, the smallest window size would be optimal in tracking the non-constant statistics of the noise power across the OFDM sub-carriers. In the case where the noise has both white and colored components, an optimal window sizes need to be selected to have an estimation error trade-off between both noise components and provide more reliable estimates of the noise power across the OFDM sub-carriers. The following sections describe the approach taken to derive the MSE criterion for the optimal window size, taking into account the power of the white and colored noise, as well as the variation of the combined noise across the OFDM sub-carriers.

4.4.4 Auto-Regressive (AR) Modeling of Colored Noise

The colored interference can be modeled as an auto-regressive (AR) process in time domain. By definition, a process is considered to be auto-regressive if it satisfies the following difference equation,

$$\boxed{c_n = \phi_1 c_{n-1} + \phi_2 c_{n-2} + \ldots + \phi_p c_{n-p} + \epsilon_n} \tag{4.5}$$

where $\phi_1, \phi_2, \ldots, \phi_p$ are the AR coefficients and ϵ_n is uncorrelated white noise with zero mean and variance σ_n^2 [48]. The interference can also be looked at as another OFDM symbol passed through a frequency selective fading channel, which is an Infinite Impulse Response (IIR) filter whose coefficients are $\phi_1, \phi_2, \ldots, \phi_p$. The model order of the process is directly proportional to the degree of power spectral variation of the AR model. The power spectrum of an AR process can be estimated using the following expression,

$$\boxed{P_Z(k) = \frac{\sigma_n^2}{|1 - \sum_{a=1}^{p} \phi_p e^{(-2\pi jka)}|^2}} \tag{4.6}$$

4.4.5 Mean-Squared-Error of Windowing

The interfering noise is viewed as a time domain signal given as,

$$z_n = c_n + n_n \tag{4.7}$$

As mentioned, the FFT of the time domain signal z_n is Z_k, which comprises of the pure colored component C_k and N_k which are the FFT of c_n and n_n respectively. By averaging the absolute squares of the process Z_k across a sliding window of size W as shown in Fig. 4.5., the MSE of the estimator within each window of width W is given by,

$$MSE_{est} = E\Big[\Big(\frac{1}{W}\sum_{l=k-w/2}^{k+w/2} Z_l^2 - (\sigma_0^2 + C_k^2)\Big)^2\Big] \tag{4.8}$$

where $Z_l^2 = (C_l + n_l)^2$ is the combined white and colored noise power which is averaged within the sliding window, σ_0^2 is the variance of the white noise and C_k^2 is the sub-carrier indexed colored noise power.

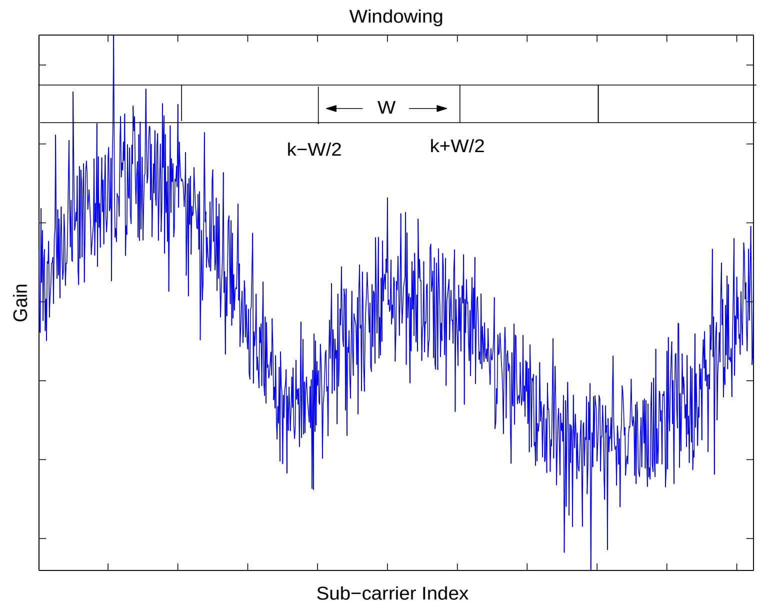

Figure 4.5. Illustration of Frequency Domain Windowing of White plus Colored Noise.

Upon substitution and simple rearrangement of terms, Eq. (4.8) can be rewritten as,

$$MSE_{est} = E\Big[\Big(\frac{1}{W}\sum_{l=k-w/2}^{k+w/2}(C_l+n_l)^2-(\sigma_0^2+C_k^2)\Big)^2\Big] \tag{4.9}$$

$$MSE_{est} = E\Big[\Big(\frac{1}{W}\sum_{l=k-w/2}^{k+w/2}(C_l^2+n_l^2+2C_ln_l)-(\sigma_0^2+C_k^2)\Big)^2\Big] \tag{4.10}$$

$$MSE_{est} = E\Bigg\{\Bigg[\underbrace{\Big(\frac{1}{W}\sum_{l=k-w/2}^{k+w/2}C_l^2-C_i^2\Big)}_{A}+\underbrace{\Big(\frac{1}{W}\sum_{l=k-w/2}^{k+w/2}n_l^2-\sigma_0^2\Big)}_{B}+\underbrace{\Big(\frac{1}{W}\sum_{l=k-w/2}^{k+w/2}C_ln_l\Big)}_{C}\Bigg]^2\Bigg\} \tag{4.11}$$

$$MSE_{est} = E\{[A^2+B^2+C^2+2AB+2BC+2AC]\} \tag{4.12}$$

Expanding and simplifying the individual terms we get,

$$\begin{aligned} E[A^2] &= E\Big[\frac{1}{W^2}\sum_{u=k-w/2}^{k+w/2}\sum_{v=k-w/2}^{k+w/2}C_u^2C_v^2-\frac{2}{W}C_k^2\sum_{l=k-w/2}^{k+w/2}C_l^2+C_k^4\Big] \\ &= \frac{1}{W^2}\sum_{u=k-w/2}^{k+w/2}\sum_{v=k-w/2}^{k+w/2}R_{C^2}(u,v)-\frac{2}{W}\sum_{l=k-w/2}^{k+w/2}R_{C^2}(l-k)+R_{C_k^4}(0) \\ &= \frac{R_{C_l^2}(0)}{W}+\frac{2}{W^2}\sum_{j=1}^{W-1}(W-j)R_{C_l^2}(j)-\frac{2}{W}\sum_{p=-w/2}^{w/2}R_{C_l^2}(p)+R_{C^4}(0) \end{aligned} \tag{4.13}$$

$$E[B^2] = E\left[\Big(\frac{1}{W}\sum_{l=k-w/2}^{k+w/2}n_l^2-\sigma_0^2\Big)^2\right] = \frac{2(\sigma_0^2)^2}{W} \tag{4.14}$$

$$E[C^2] = \frac{4}{W^2}E\left[\Big(\sum_{l=k-w/2}^{k+w/2}C_ln_l\Big)^2\right] = \frac{4\sigma_0^2}{W^2}\sum_{l=k-w/2}^{k+w/2}R_{C_l^2}(0) \tag{4.15}$$

$$E[2AB] = 0 \tag{4.16}$$

$$E[2BC] = 0 \tag{4.17}$$

$$E[2AC] = 0 \tag{4.18}$$

Therefore, the final expression for the MSE of the noise power estimator is given by,

$$MSE_{est} = \frac{R_{C_l^2}(0)}{W} + \frac{2}{W^2}\sum_{j=1}^{W-1}(W-j)R_{C_l^2}(j) - \frac{2}{W}\sum_{p=-w/2}^{w/2} R_{C_l^2}(p)$$
$$+R_{C_l^4}(0) + \frac{2(\sigma_0^2)^2}{W} + \frac{4\sigma_0^2}{W^2}\sum_{l=k-w/2}^{k+w/2} R_{C_l^2}(0) \qquad (4.19)$$

As observed from the above expression, the MSE of the estimator takes into account the correlation of the power spectrum of the colored noise for different lags within the window, the variance of the white Gaussian noise and the size of the window, *i.e.* size of the window.

4.4.6 A Sub-optimal Approach

The optimal method to track the noise statistics would be to use a sliding windowed approach across the OFDM band, where the window size would change adaptively based on the variation of the noise statistics as shown in Fig. 4.6.. The optimal window size would be the size where the MSE is minimized.

As observed, the window size which yields the lowest error depends on the variation of the noise statistics across the OFDM sub-carriers. The size of the window should be chosen such that the MSE for that window size is minimum. It is for window sizes greater or lesser than the optimal window size that the MSE value increases, as can be seen in the adjoining MSE plot in Fig. 4.6.. The optimal window sizes are different for groups of sub-carriers and are indicated with arrows in the MSE plot.

However, in the case of sub-band adaptive modulation, this optimal approach may not be feasible due to the fact that it requires changing the size of each sub-carrier group. This would result in increased overhead information and complexity. A sub-optimal method is therefore developed with the sub-band adaptive OFDM system structure in mind. Instead of adaptively changing the window size along the sub-carriers, the average overall MSE, derived from Eq. (4.19), based on a set of fixed window sizes is minimized. The window size for which the least MSE is observed is selected. The most convenient set of window sizes would be $\{2^0, 2^1, 2^2, \ldots, 2^{log_2 N}\}$, where N is the total number of sub-carriers.

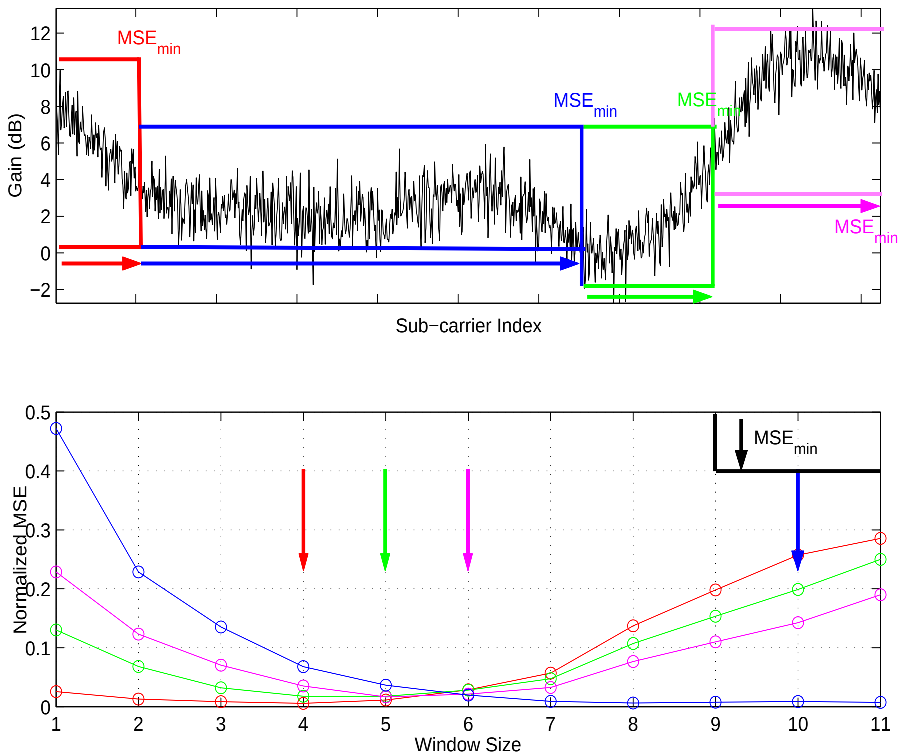

Figure 4.6. Optimal Window Size Selection Across OFDM Band.

4.5 Performance Results

The performance results are obtained through computer simulations. An OFDM system with 1024 sub-carriers is considered and the window sizes are varied from 2^0 to 2^{10}. For the colored noise, a co-channel interference source, which is modeled as an AR process of varying model orders, is used. Additive white noise is included apart from colored noise, and performance evaluations are made for white and colored noise dominated environments, which are created by varying the colored-noise-to-white-interference ratios (CINR). The theoretical curves obtained from Eq. (4.19) and practical curves are plotted for CINR values of 3 dB and -3 dB, and are found to match. As observed from Fig. 4.7., the optimal window size for a combination of colored and white noise depends on the correlation of the power spectral density for different lags within the window of the colored noise, as well as the noise variance of the white noise (σ_0^2) and window size W. As seen, the MSE decreases till it reaches the optimal window size, and then continues to increase.

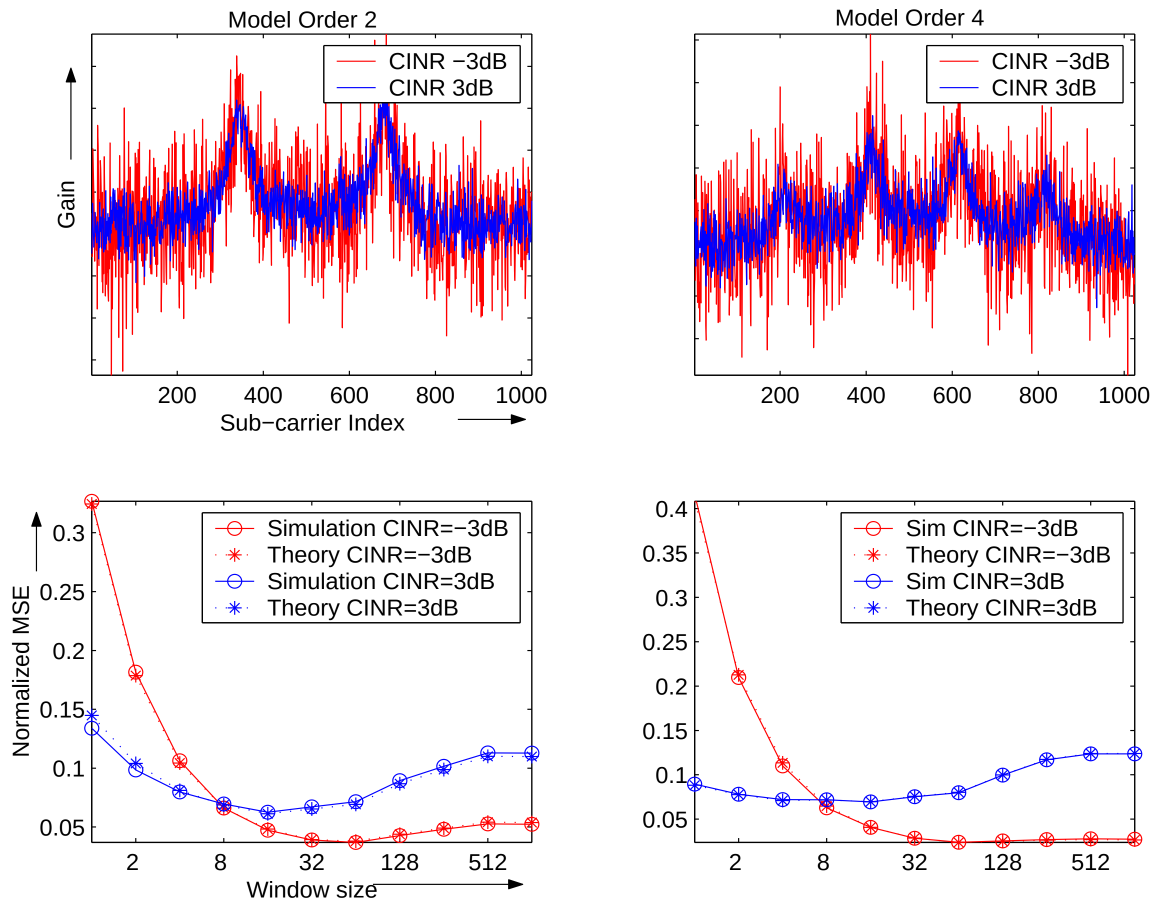

Figure 4.7. Simulation Results of Appropriate Window Size.

In colored noise dominated environments, as the model order of the colored noise defined by the AR process increases, the variation of its power spectrum in the frequency domain is more rapid due to this increase in correlation. This is reflected by the selection of a smaller window size in a colored noise dominated environment which is modeled as an AR process of model order 4 as compared with a model order 2, for the same white noise variance. This allows better tracking of the noise statistics in different sub-carrier groups and ensures optimal modulation mode selection within each sub-band. For white noise dominated environment, this estimation method reduces to the conventional technique of noise power estimation, where it is assumed that the interfering noise is completely white.

4.6 Conclusion

In this chapter, noise power estimation that removes the white noise assumption is described. The proposed algorithm considers a more practical environment where noise is characterized by a non-constant spectral content over the OFDM sub-carriers. This is often the case when noise is dominated by a strong interferer. The autocorrelation of the power spectral density (PSD) reflects the intensity of color. On the other hand, when the noise is not colored, the algorithm performs as well as the conventional noise variance estimations which are designed for white noise dominated scenarios. The proposed solution can be very useful for sub-band adaptive modulation as well as for other adaptive transmitter and receiver algorithms, like optimal soft information calculation, improved channel estimation etc.

CHAPTER 5

BLIND MODULATION MODE DETECTION

Blind modulation mode detection is a rapidly emerging field of signal exploitation because it provides information about the origin and properties of an unknown signal. Modulation detection is used predominantly for wireless communications in areas such as electronic surveillance, interference detection and countermeasures, spectrum management, etc. Military applications of modulation recognition are more prevalent since efficient jamming can be carried out. Currently, the most significant application is in the area of software radio and other reconfigurable communication systems. In light of adaptive and sub-band adaptive OFDM systems, efficient blind modulation mode detection further improves throughput, data capacity and spectral efficiency of the system because it eliminates the need to transmit explicit modulation information. The gain in data capacity can potentially be used for other purposes such as additional data transmission, PAPR reduction, etc.

Following an overview of the concept and past work in blind modulation detection, this chapter presents an improved statistical blind modulation mode detection algorithm using the Kullback-Leibler (K-L) distance [49].[1] The approach is based on exploiting the minimum Euclidean distances between the received samples and the nearest legitimate constellation points of all possible modulation modes. At low SNR values, previously proposed methods of blind modulation detection in adaptive OFDM systems suffered a bias toward the higher modulation modes. The proposed statistical algorithm eliminates this bias while still maintaining an acceptable amount of computational complexity.

[1] This work is partly published in[6] and it is currently under review for another publication[7].

5.1 Literature Overview

Blind modulation detection has been traditionally approached in two ways, namely a pattern recognition and a decision theoretic approach [50]. Pattern recognition attempts to extract a unique feature that is typical of a particular modulation format for use in a statistical classifier. Several recognized pattern recognition algorithms, such as Fuzzy C-means clustering [51], have been proposed and tested to perform well, provided that there are a relatively large number of samples available to make a decision. Within the scope of adaptive OFDM systems, there are no algorithms in literature to the authors knowledge which follow a pattern recognition approach till date. This is primarily because it is desired that detection be done in real time with a low number of samples. A large sample size would require excessive memory, and hence signaling is considered a better alternative as opposed to using the pattern recognition approach.

The objective of blind modulation detection is to determine the type of modulation used by the least possible number of received samples. The only empirical data provided by the received noisy samples is the error distance to the closest legitimate constellation points of all possible modulation schemes by the transmitter. In other words, given a noisy sample, there would be M errors where M is the number of modulation schemes used. Therefore, the objective is to make use of the distribution of this empirical data or errors, to make a statistical inference of the type of modulation used.

Two methods with reference to blind modulation detection for adaptive OFDM systems are proposed in [5], which are based on the decision theoretic approach. In the first method, the mean Euclidean distance between the received samples and all the closest legitimate constellation points of all possible modulation schemes are calculated. The average Euclidean distance for different hypotheses can be calculated as,

$$\boxed{e_m = \frac{\sum_s |R_{s,m} - \hat{R}_{s,m}|^2}{S} \quad m = BPSK,\ QPSK\ etc.} \tag{5.1}$$

where R is the received sample, $\hat{R}$ is the hypothesis of the received sample, m is the index of the modulation, and S is the number of samples used for averaging. The scheme which minimizes the average Euclidean distance e in Eq. (5.1) is the chosen for demodulation. However, there is always a bias towards the higher order modulation schemes irrespective of the actual modulation used. This is because there are more number of legitimate constellation points for higher order modulation schemes, and hence Eq. (5.1) is always minimal for the highest modulation mode at low SNR values. The second method exploits the error correction capability of the turbo codec in terms of its input and corresponding output bit probability. Since coding is not used within the scope of this thesis, this method is not used as a basis for performance comparison.

5.2 A Model Selection Approach

Modulation detection is treated here as a model selection problem. The empirical data provided by the noisy received samples is used to construct a set of probability distribution functions (PDFs), which is referred to as the model. This model is approximated to a set of known candidate models obtained beforehand, and the best approximation would yield the modulation information of the received samples. The metric used to approximate the models is the *Kullback-Leibler (K-L) Information* .

Well over a century ago, measures were derived for assessing the 'distance' between two models or PDFs. The most relevant here is the Boltzmann's (1877) concept of generalized entropy, and Shannon's (1948) famous treatise on communications theory. Kullback and Leibler (1951) derived an information measure that happened to be the negative of Boltzmann's entropy, now referred to as the Kullback-Leibler or K-L distance. It can be conceptualized as a measure of discrepancy between two models, and not a simple measure of distance. The K-L distance is perhaps the most fundamental of all information measures in the sense that it is derived from minimal assumptions and is the logical basis for model selection in conjunction with likelihood inference.

The K-L distance is always positive, except when the two distributions are identical, in which case it would be zero. Details of the derivation and formulation can be found in [52].

5.3 Kullback-Leibler Distance Algorithm

K-L distance is the measure of discrepancy or information loss when one PDF is approximated to another [52]. Model selection aims at selecting a candidate model, among many, that minimizes this information loss. In forthcoming discussions, the PDFs obtained from the received samples will be referred to as P and that of the approximating model, a known distribution to which P is approximated to, as Q. The expression for the K-L distance between two discrete PDFs is,

$$\boxed{KL(P,Q) = \sum_{b=1}^{B} P_b \ \log\left[\frac{P_b}{Q_b}\right]} \tag{5.2}$$

where B is the total number of bins in the PDF. When multiple candidate models are used, the model which minimizes $KL\big(P,Q(l)\big)$ is chosen as the best directional fit between what is observed and all the candidate models indexed by l. The minimum Euclidean distances between the received noisy sample and the M constellation points corresponding to each modulation scheme is given as,

$$\boxed{e_m(k) = \left|\frac{Y(k)}{\hat{H}(k)} - \tilde{X}_m(k)\right|^2 \qquad 1 \leq m \leq M} \tag{5.3}$$

where $Y(k)$ is the received sample on the k^{th} subcarrier, $\hat{H}(k)$ is the estimated channel frequency response, and $\hat{X}_m(k)$ is the closest hypothesized constellation point of the modulation scheme indexed by m. Therefore, from a block of received noisy samples, a set of M Euclidean distance distributions corresponding to each modulation scheme can be obtained using Eq. (5.3). Their respective PDFs, *i.e.* $p_1, p_2, \ldots, p_M$, are collectively referred to as the model P. If the candidate models for different modulation schemes at different SNRs were known beforehand, the minimization of the K-L distances between the PDFs constituting P, and the their corresponding PDFs in each of the candidate models, would point to a particular modulation scheme. One such candidate model is illustrated in Fig. 5.1. for BPSK modulated symbols at channel SNR of 5 dB.

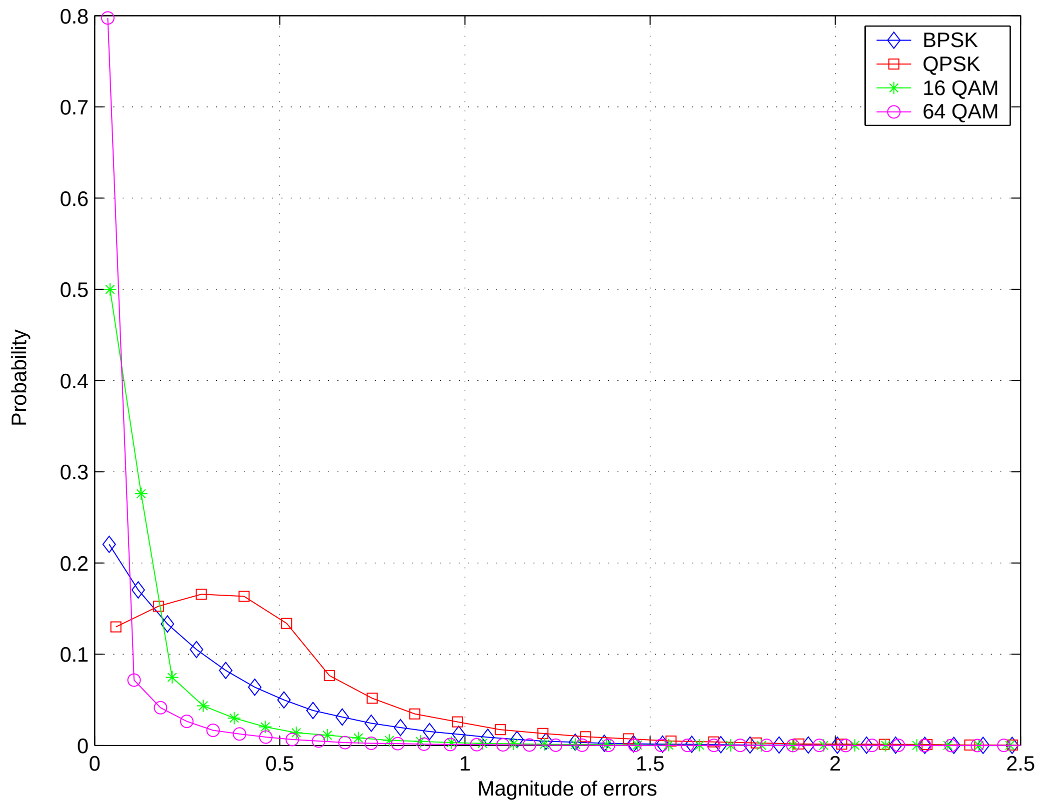

Figure 5.1. Probability Distribution of Errors with BPSK at Channel SNR of 5 dB.

A reliable set of PDFs constituting P requires a large number of samples to be used. On the other hand, it is desirable that the number of samples used to construct P be low in order to reduce computational complexity, while still maintaining an acceptable level of performance. Hence, as a trade-off, only the cumulative K-L distances between the corresponding first order moments of distributions in P and the candidate models are calculated and minimized. The dependance of the first order statistics on the modulation scheme used and SNR is illustrated by their distribution in Fig. 5.2., for different modulation schemes at same SNRs and vice versa.

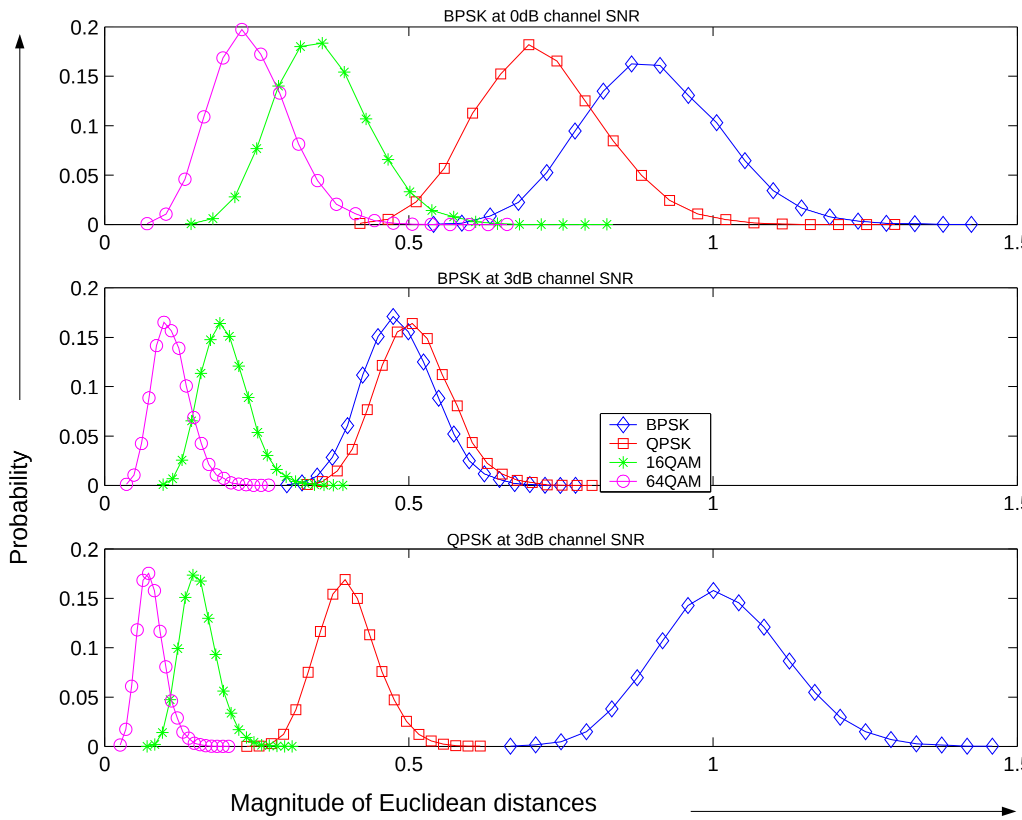

Figure 5.2. Effect of the Number of Samples on the First Order Statistics of the Probability Distribution of Errors.

5.4 Algorithm

5.4.1 Detection Based on Model Selection

Let us assume that the first order statistics of the PDFs at each integer SNR value between limits s_1 and s_2 are completely known when each of the M modulation schemes are used. It is desirable that the limits s_1 and s_2 span all possible SNR values that are characteristic of the system at hand. This set of candidate models is represented by $Q(m, s)$, where $s_1 \leq s \leq s_2$ and $1 \leq m \leq M$. From a received block of noisy samples, the distributions $p_1, p_2, \ldots, p_M$ corresponding to each modulation scheme is obtained, *i.e.* model P. The cumulative K-L distances between the first order statistics of P and each of the known candidate models is calculated and minimized. For each OFDM block, it is written as,

$$\boxed{KL\big(P, Q(m,s)\big) = \sum_{m=1}^{M} E\big(p_m\big) log \left[\frac{E\big(p_m\big)}{E\big(q_m\big)} \right]} \tag{5.4}$$

Modulation detection is thereby converted into a multiple hypothesis testing problem for integer SNR value between s_1 and s_2. The modulation mode m for which the K-L distance is least is selected for demodulation.

5.4.2 Detection Based on SNR Estimation

In situations where the transmitter and receiver have a priori knowledge of the SNR bounds within which every modulation scheme is used, a slight variation to the model selection technique is proposed. It is based on the idea that if the receiver can estimate the approximate SNR at the transmitter using the received samples, it can estimate the modulation scheme used with the knowledge of the SNR bounds. Modulation schemes at the transmitter are selected such that a BER of 10^{-3} is achieved and the SNR bounds within which each scheme is used are shown in Fig. 5.3..

Therefore, for each SNR value s, between limits s_1 and s_2, there is only one modulation scheme used and hence only one candidate model. Therefore, Eq. (5.4) can be re-written

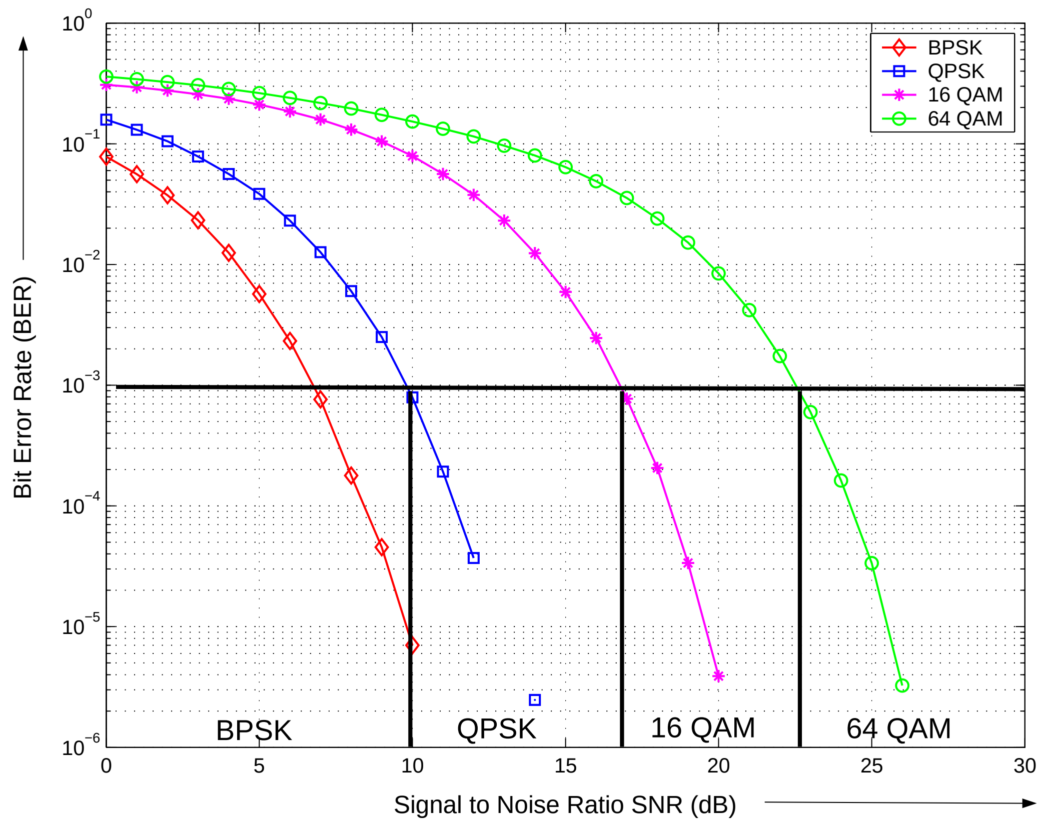

Figure 5.3. SNR Bounds of Different Schemes to Achieve 10^{-3} BER.

as follows,

$$\boxed{KL\big(P, Q(s)\big) = \sum_{m=1}^{M} E(p_m) log \left[\frac{E(p_m)}{E(q_m)}\right] \quad s_1 \leq s \leq s_2} \tag{5.5}$$

By minimizing the K-L distance between the first order statistics of p and that of all the candidate models, the approximate SNR is found. It is then compared against a look-up table to estimate the modulation scheme used at that SNR. The underlying assumption here is that the transmitter selects the modulation schemes perfectly and the noise variance between the transmitter and receiver does not differ significantly. In practical applications, the transmitter and receiver would suffer from different noise levels. Simulation results are presented for different levels of noise mismatch modeled by,

$$\boxed{SNR_{Rx} = SNR_{Tx} \pm \zeta} \tag{5.6}$$

where SNR_{Rx} and SNR_{Tx} are the SNRs at the receiver and transmitter respectively and ζ is the random mismatch between them which is modeled by a uniform distribution between $\pm\rho$, where ρ is the maximum specified SNR mismatch in dB.

5.5 Results

The proposed algorithm is tested for an OFDM system with 64 sub-carriers. Four modulation schemes namely, BPSK, QPSK, 16-QAM and 64-QAM are used for transmission. The SNR bounds within which the modulation schemes are selected is based on an achievable 10^{-3} BER criterion, as shown in Fig. 5.3.. For simulations, perfect channel knowledge and an additive white Gaussian noise channel are assumed.

As observed in Fig. 5.4., the algorithm based on minimal Euclidean distance presented in [5] yields a detection error rate (DER) of 10^{-3} only above channel SNR values of 18 dB. Detection based on the proposed model selection algorithm is observed to perform exceptionally well in comparison, and achieves the same DER at 3.8 dB channel SNR.

Detection by SNR estimation performs without any errors when noise levels are the same at the transmitter and receiver. For practical cases, this assumption has been removed and the robustness of this algorithm is tested for ρ values of 2 dB, 3 dB, 4 dB and 5 dB.

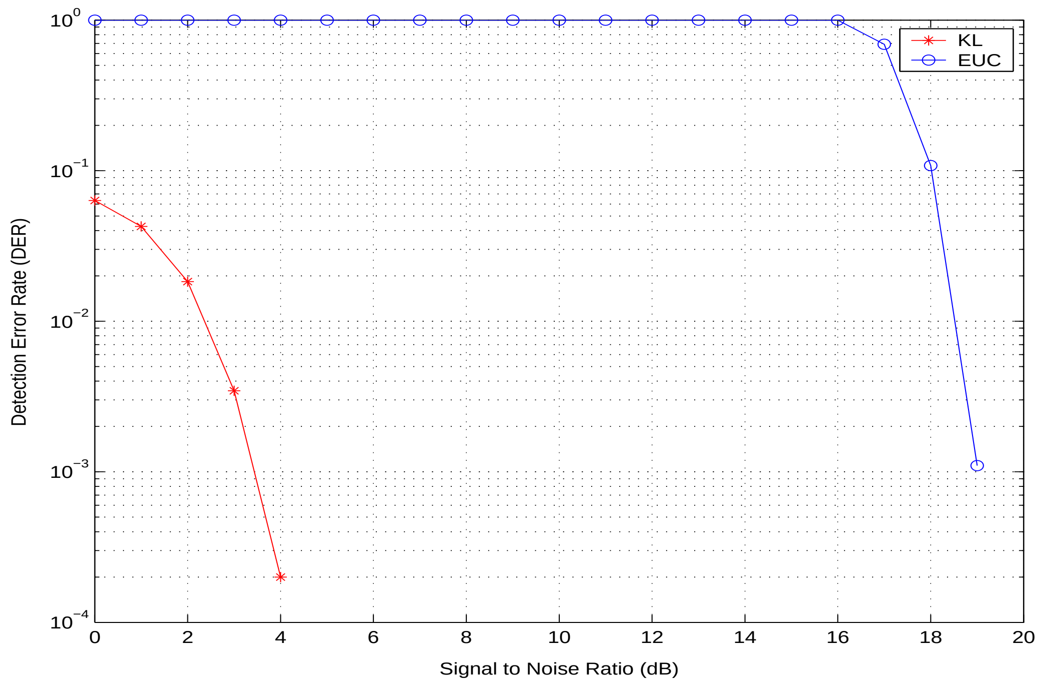

Figure 5.4. Detection Error Rate Comparison Between the Euclidean, MSE and K-L Algorithms. No Incorrect Decisions with K-L Algorithm for Sample Size of 64 and Equal Noise Variance at the Transmitter and Receiver.

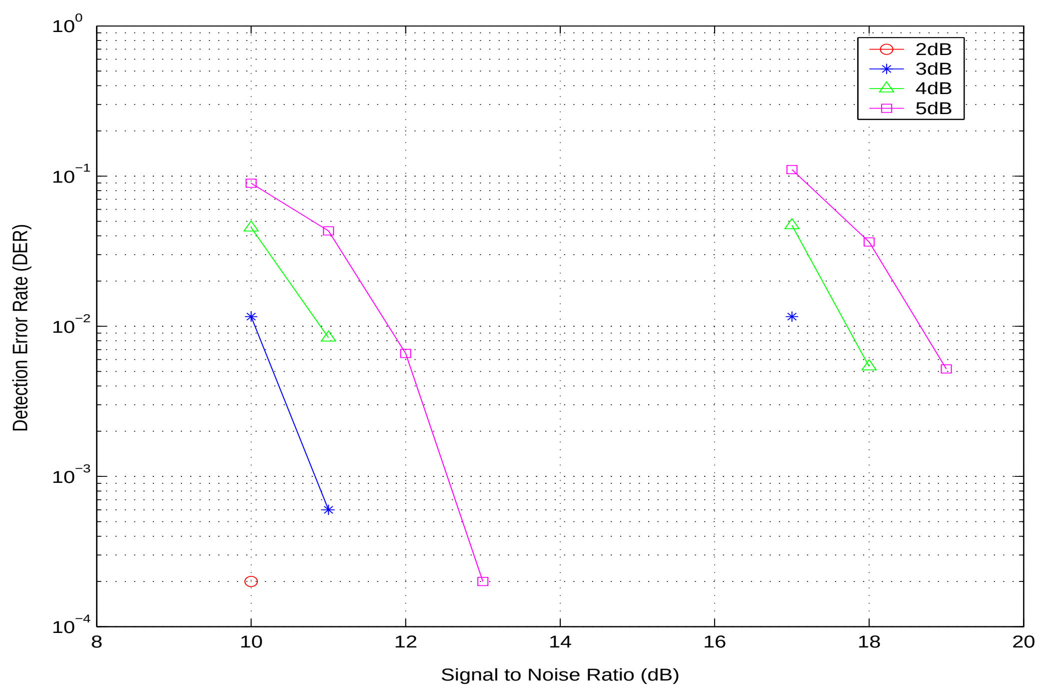

Figure 5.5. Detection Error Rate Comparison between the Euclidean Distance Minimization and Detection by Model Selection Algorithms.

Results in Fig. 5.5. show prominent errors at the modulation transition regions of neighboring modulation schemes. This is expected because a noise power mismatch would lead to a selection of the wrong modulation scheme at the transmitter, the effect of which is more pronounced at the transition regions of neighboring modulation schemes rather than between the bounds of each modulation scheme.

5.6 Conclusion

In this chapter, blind modulation detection in reference to adaptive and sub-band adaptive OFDM based wireless systems was investigated. A more efficient statistical blind modulation detection technique based on the K-L model selection criterion was presented. The approach was to use the empirical data provided by the noisy received samples in order to estimate the modulation scheme used by the transmitter. The PDF of the Euclidian distances or errors from the received samples are compared to a set of known distributions and the closest match would yield the modulation mode used.

The algorithm was tested under known channel conditions in the presence of additive Gaussian noise and practical imperfections such as noise mismatch between transmitter and receiver. Performance results using Monte Carlo simulations were presented and were found to significantly better at lower SNR values than previously proposed methods, without an increase in complexity.

CHAPTER 6

CONCLUSION AND FUTURE RESEARCH

Research on OFDM has been progressing by leaps and bounds during the recent past, which only goes to reflect its widespread recognition in next generation high data-rate applications. Its ease of implementation and the ability to mitigate ISI has sparked its acceptance as a physical layer standard in a wide range of systems from Terrestrial Digital Video Broadcasting in Europe to WLAN systems in Europe, USA and Japan. Variable-mode or adaptive systems have also been in the technological forefront because they provide the best compromise among a number of contradicting design factors such as spectral efficiency, channel capacity, etc. Adaptive systems in conjunction with OFDM provide an excellent way to satisfy the requirements posed by next generation wireless applications and services. In times where spectrum is scarce and the need for maximal utilization of available resources is high, adaptive OFDM systems offer a means to provide attractive performance with acceptable level of complexity.

This thesis has investigated efficient parameter estimation and blind modulation detection techniques for use in adaptive and sub-band adaptive OFDM wireless systems. Conventional techniques cannot provide optimal performance simply due to the fact that they are not tailored to adaptive OFDM systems. Efficient algorithms in the areas of noise power and SNR estimation, and blind modulation mode detection that are specific to adaptive OFDM systems are devised, keeping in mind the practicality of implementation.

In noise power estimation, the conventional assumption that the interfering noise is white and Gaussian distributed is removed. In real world scenarios, the interfering noise is a combination of colored and white noise. Sub-band adaptive modulation systems require a good estimate of the variation of noise statistics within the OFDM band. With this as a motivating factor, A novel frequency domain windowing technique that takes into account this power spectral variation of the noise across the OFDM sub-carriers is proposed. This has been found to improve noise power estimation dramatically, with a negligible increase in complexity. Much can be done to further enhance the performance of this method. Time domain windowing that takes into account the time varying nature of the mobile channel can be considered as a further research topic. The optimal window size both in time and frequency will have the capability to follow the noise statistics in both domains to optimize the performance of the adaptive OFDM system. Time domain variation of the channel for mobile OFDM systems can also be taken into consideration to drive a two dimensional optimal window size. Although this would be a straightforward extension of our proposed method, the results would be interesting to many researchers.

Sub-band adaptive OFDM systems change the modulation modes within different groups of sub-carriers of the OFDM symbol. Signaling this information for proper demodulation at the receiver on reserved sub-carriers decreases the data capacity and throughput of the system. Blind modulation mode detection is an efficient alternative to circumvent this problem. A computationally simple statistical blind modulation detection algorithm based on the K-L distance is presented, and has been found to work well at low channel SNR values. Aspects of this algorithm that need further thought are its performance in the presence of narrow-band interference and more importantly, channel estimation errors. The modulation modes are selected based on a required BER criterion, however possible system performance gains may be achieved by dynamically choosing the modulation technique based on the type of data being transmitted.

In conclusion, adaptive OFDM promises to be a good choice of technology that can be depended upon for high speed wireless networks. Ongoing research on aspects of adaptivity guarantees to deliver a high quality of user experience in the multimedia-centric future of wireless communications.

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